

MyPBX

U300

Administrator's Guide

Version 12.17.0.30

Yeastar Technology Co., Ltd.

Table of Contents

1. Introduction	5
1.1 Features	6
1.2 Hardware Specifications.....	7
1.2.1 Exterior Appearance	7
2. System setup	7
2.1 Connection Drawing	8
2.2 Connecting Ethernet Line.....	8
2.3 Supplying Power	8
3 Administrator Login	9
4 Status	10
4.1 Line status	10
4.1.1 Extension Status	10
4.1.2 Trunk Status	11
4.2 System Status	11
4.2.1 System Info.....	12
4.2.2 Network Status	12
5 System	13
5.1 Network Preferences	13
5.1.1 LAN Settings.....	13
5.1.2 WAN Settings.....	14
5.1.3 DHCP Server.....	15
5.1.4 VLAN Settings.....	16
5.1.5 VPN Settings.....	19
5.1.6 DDNS Settings	19
5.1.7 Static Route.....	20
5.2 Firewall Settings	21
5.2.1 Firewall Rules.....	21
5.2.2 IP blacklist.....	23
5.3 System Preference	24
5.3.1 Password Settings	24
5.3.2 Date and Time	25
5.3.3 Firmware Update.....	26
5.3.4 Hot Standby	26
5.3.5 Backup and Restore	28
5.3.6 Reset and Reboot	28
5.3.7 Alert Settings.....	29
5.3.8 Addons	31
5.4 Storage Management	32
5.4.1 External Storage	32
6 PBX	35
6.1 Extensions	35
6.1.1 FXS/VoIP Extensions.....	35

FXS Extensions	35
VoIP Extensions	40
6.1.2 Phone Provisioning	45
6.2 Trunks	54
6.2.1 Physical Trunk	54
6.2.2 VoIP Trunk.....	58
6.3 Outbound Call Control	64
6.3.1 Outbound Routes.....	64
6.3.2 Speed Dial Settings	66
6.4 Inbound Call Control	67
6.4.1 IVR.....	67
6.4.2 Ring Groups.....	70
6.4.3 Queues	71
6.4.4 Conferences.....	75
6.4.5 Inbound Routes.....	75
6.5 Audio Settings	80
6.5.1 Custom Prompts.....	80
6.5.2 Music on Hold Prompts.....	81
6.5.3 System Prompts Settings	82
6.6 Basic Settings.....	82
6.6.1 General Preferences.....	83
6.6.2 Business Hours	85
6.6.3 Feature Codes.....	86
6.6.4 Voicemail Settings	89
6.7 Advanced Settings	92
6.7.1 SIP Settings.....	92
6.7.2 IAX Settings	98
6.7.3 Blacklist	99
6.7.4 Callback Settings.....	100
6.7.5 DNIS Settings	101
6.7.6 DISA.....	102
6.7.5 PIN User Settings	103
6.7.8 Paging Groups.....	105
7 Reports	107
7.1 Call Logs.....	107
7.2 System Logs	107
8 Logout	108
9. Use MyPBX	109
9.1 Make outbound call	109
9.1.1 Sample Routing via E1 Trunk	109
9.1.2 Sample Routing via VoIP Trunk.....	110
9.2 Incoming call.....	112
9.2.1 Sample Routing to an IVR.....	112

APPENDIX A FAQ	115
APPENDIX B How to Configure Autobackup	116
APPENDIX C How to Configure NAT setting	118
APPENDIX D How to Use Auto Provision	120
APPENDIX E HowDo I Configure Distinctive Ring Tones	123
APPENDIX F How to Use DID	125
APPENDIX G MyPBX Security configuration Guide	129
1. Security Configuration for Web GUI.....	129
1.1 Change the default access port for HTTP on Options page	129
1.2 Change the default password for the web GUI	130
2. Disable SSH on LAN Settings Page	130
2.1 Disable SSH.....	130
2.2 Change the default password for SSH	131
3. Security Configuration for Extensions	132
3.1 Change the default SIP Port.....	132
3.2*Disable guest calls.....	133
3.3* Security Configuration for remote extensions	134
3.4 Set an enhanced password and enable IP restriction for extensions	134
4. Setup Proper Firewall Rules.....	135
5*. Alert Settings	142
5.1 IPATTACK	143
5.2 WEBLOGIN	144
6. Note	145

1. Introduction

MyPBX –IP-PBX for Medium Businesses/Home Office

MyPBX U300 is a standalone embedded hybrid PBX for medium businesses and remote branch offices of larger organizations (1-100 users per site). MyPBX U300 also offers a hybrid solution (a combination of VoIP applications using E1/T1/J1 equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

FCC Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radiofrequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

FCC Caution

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

FCC Warning

This equipment has been tested and found to comply with the regulations for a Class B digital device, pursuant to Part 15 of the FCC Rules.

Operation is subject to the following two conditions:

- (1) This device may not cause harmful interference.
- (2) This device must accept any interference received including interference that may cause undesired operation.

1.1 Features

• Auto-provision	• Follow me
• Audio In/Out	• Interactive Voice Response (IVR)
• BLF Support	• Intercom / Zone Intercom
• Blacklist	• Music On Hold
• Call transfer	• Music On Transfer
• Call Detail Records(CDR)	• Paging / Zone Paging
• Call Forward	• PIN Users
• Call Parking	• Queue
• Call Recording	• QOS
• Call Pickup	• Ring Group
• Call Routing	• Route by Caller ID
• Call Transfer	• Spy functions
• Call Waiting	• Skype Integration (Skype Connect)
• Caller ID	• Three-way Calling
• Call Back	• Mobility Extension
• Conference	• External Storage
• Speed Dial	• DDNS
• Define Office Time	• OpenVPN
• Direct Inward System Access(DISA)	• T.38
• DIDs	• Voicemail
• Distinctive Ringtone	• VLAN
• Do Not Disturb(DND)	• WAN
• Dial by Name	• PPPoE
• Firewalls	• Static Route
• Alert	• Multiple administrators
• Integrated built-in packet capture tools	• Blind Transfer

More info, please click: <http://www.yeastar.com/Products/MyPBX-U300.asp>

1.2 Hardware Specifications

1.2.1 Exterior Appearance

Front Panel



Figure1-1 MyPBX U300 Front Panel

No.	Indication
①	Reset button: Press until all LEDs turn orange, which means the reset is completed.
②	Power LED: Green stably Run LED: Green Blinking
③	Audio in interface Audio out interface
④	USB 2.0 port
⑤	WAN port LAN port
⑥	Console port (RJ45)
⑦	E1/T1 interface
⑧⑨	Two fixed FXS ports(RJ11)

2. System setup

2.1 Connection Drawing

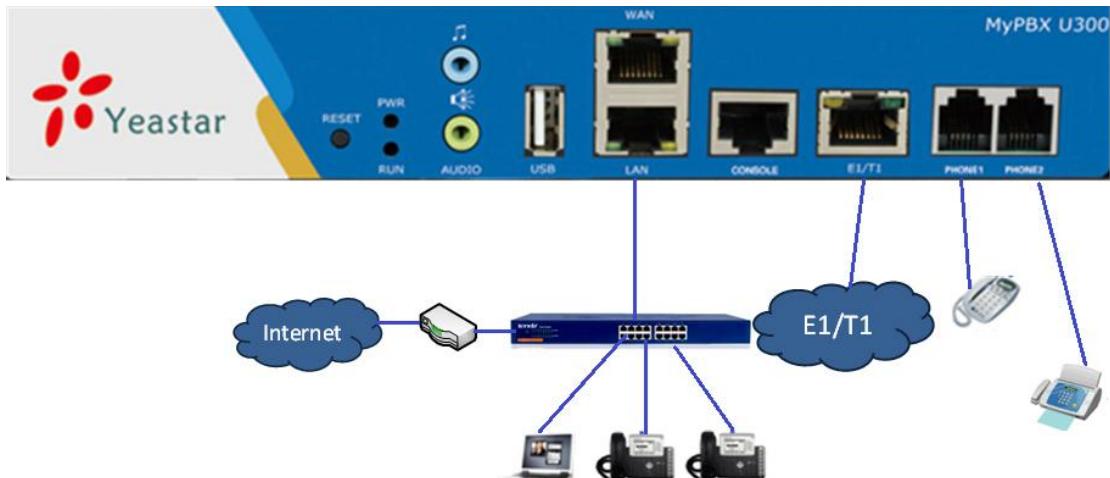


Figure 2-1

2.2 Connecting Ethernet Line

MyPBX provides two 10/100MEthernet ports with RJ45 interface and LED indicator. Plug Ethernet line into MyPBX's Ethernet port, and then connect the other end of the Ethernet line with a hub, switch, router, LAN or WAN. Once connected, check the status of the LED indicator. A yellow LED indicates the port is in the connection process, and a green LED indicates the port is properly connected.

2.3 Supplying Power

Please follow the steps below to connect the MyPBX unit to a power outlet:

1. Connect the small end of the power cable to the power input port on the MyPBX back panel, and plug the other end of the cable into a 100V~240V AC power outlet.
2. Check the Power LED on the front panel. A solid green LED indicates that power is being supplied correctly.

3 Administrator Login

From your web browser, input the IP address of the MyPBX server.

If this is the first time you are configuring MyPBX, please use the default settings below (your PC should be in the same local network with MyPBX):

IP Address: <http://192.168.5.150>

Username: **admin**

Password: **password**

In this example, the IP address is 192.168.5.193



Figure 3-1

This is the welcome page of MyPBX U300 after successful login.

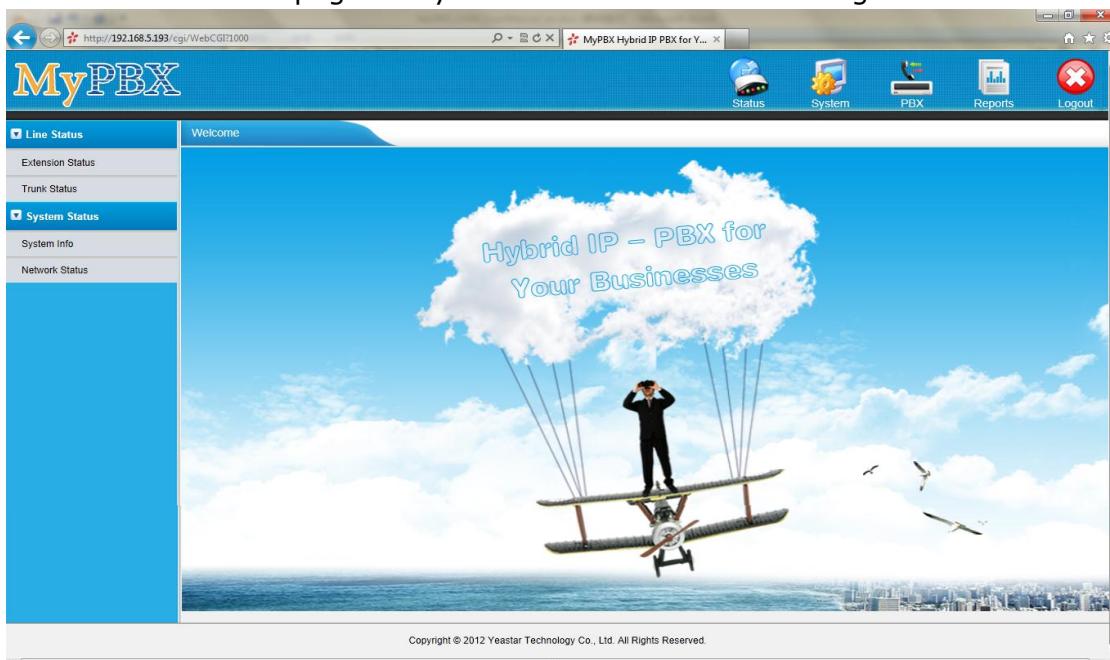


Figure 3-2

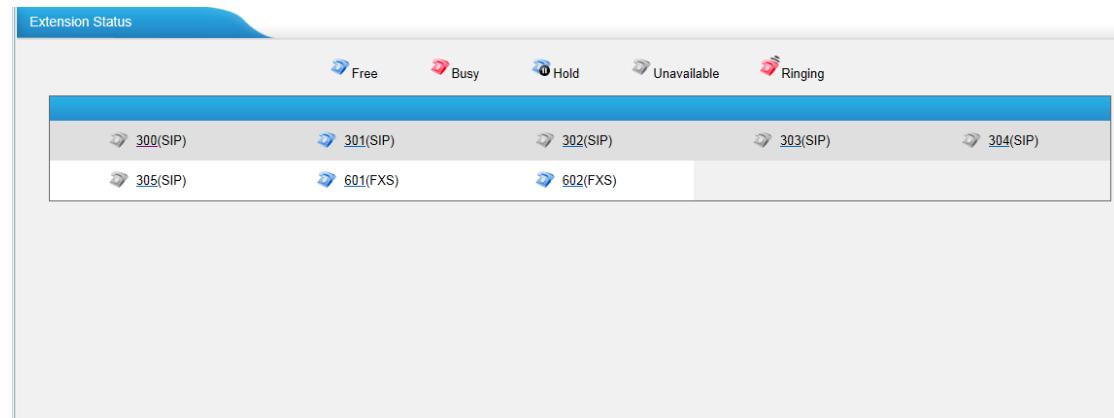
4 Status

Click  to start to check the status of MyPBX U300, where we can check the status of extension, trunk, network and system information.

4.1 Line status

In this page, we can check the status of extension and trunks

4.1.1 Extension Status



The screenshot shows a user interface titled "Extension Status". At the top, there are five status icons with labels: "Free" (green phone), "Busy" (red phone), "Hold" (blue phone with a play/pause icon), "Unavailable" (grey phone), and "Ringing" (red phone with a bell icon). Below this is a grid of extension status entries. Each entry consists of a small icon followed by a label. The labels are: "300(SIP)", "301(SIP)", "302(SIP)", "303(SIP)", "304(SIP)" in the first row; and "305(SIP)", "601(FXS)", "602(FXS)" in the second row. The background of the grid rows alternates between light grey and white.

Figure 4-1

MyPBX Status Description:

Extensions:

- 1)  : Extension is unavailable
- 2)  : Extension is idle
- 3)  : Extension is ringing
- 4)  : Extension is busy
- 5)  : Extension is on hold

4.1.2 Trunk Status

Trunk Status						
Status	Signal	Trunk Name	Type	User Name	Port/Hostname/IP	Reachability
Registered		Yeastar	SIP	504	192.168.4.142	OK
OK (2 ms)		Support.Tel	SP-SIP		192.168.4.141	OK (2 ms)
Type	Connect			Alarm		
E1						

Figure 4-2

Trunks:
VOIP Trunk:
Status

Unregistered: Trunk registration failed.

Registered: Successful registration, trunk is ready for use.

Request Send: Registering.

Waiting: Waiting for authentication.

Service Provider:
Status

OK: Successful registration, trunk is ready for use.

Unreachable: The trunk is unreachable.

Failed: Trunk registration failed.

E1 Trunk:

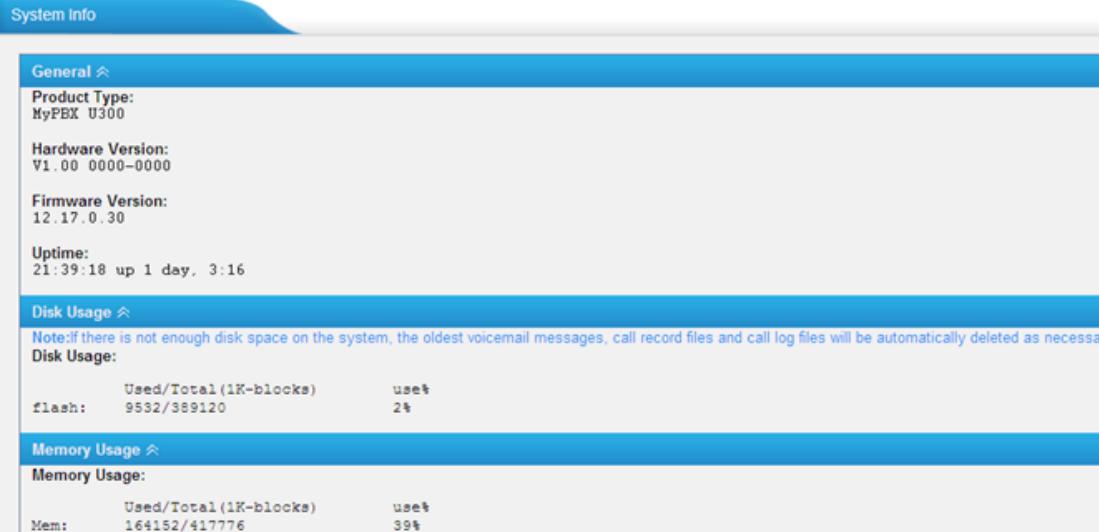
Connect	Alarm	Comment
		The port doesn't have E1 line connected.
		E1 line has been connected to the port; PRI is in corresponding or can't communicate.
		Succeed.

4.2 System Status

In this page, we can check the status of MyPBX system, including the hardware, firmware version and the network status of LAN and WAN ports .

4.2.1 System Info

In this page, we can check the hardware/firmware version, or the disk usage of MyPBX.



General

- Product Type:** MyPBX U300
- Hardware Version:** V1.00 0000-0000
- Firmware Version:** 12.17.0.30
- Uptime:** 21:39:18 up 1 day, 3:16

Disk Usage

Note: If there is not enough disk space on the system, the oldest voicemail messages, call record files and call log files will be automatically deleted as necessary.

	Used/Total(1K-blocks)	use%
flash:	9532/389120	2%

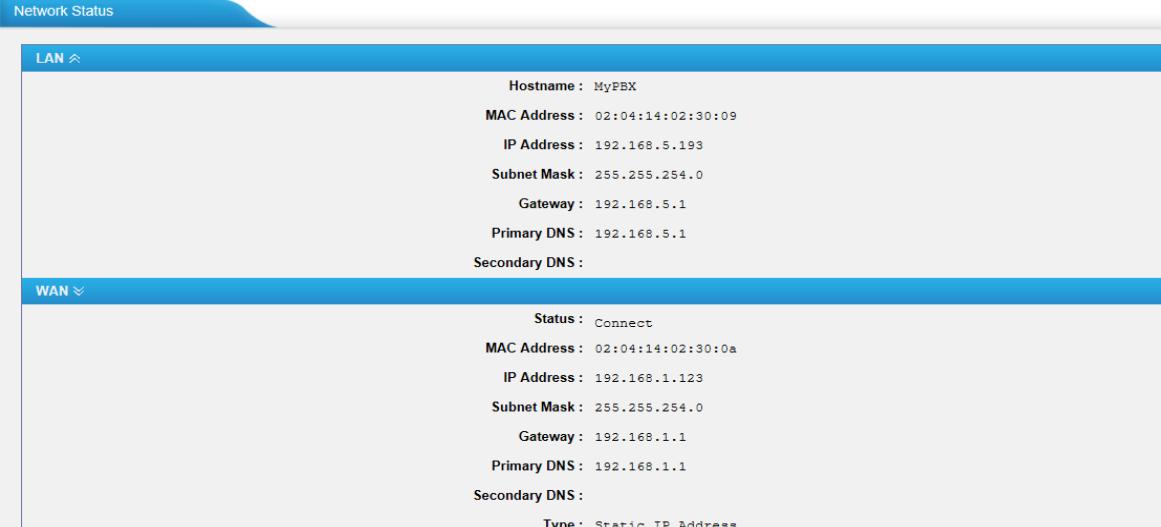
Memory Usage

	Used/Total(1K-blocks)	use%
Mem:	164152/417776	39%

Figure 4-3

4.2.2 Network Status

In this page, the IP address of LAN and WAN port will appear, if OpenVPN and Vlan are configured well, they will be display here too.



LAN

- Hostname:** MyPBX
- MAC Address:** 02:04:14:02:30:09
- IP Address:** 192.168.5.193
- Subnet Mask:** 255.255.254.0
- Gateway:** 192.168.5.1
- Primary DNS:** 192.168.5.1
- Secondary DNS:**

WAN

- Status:** Connect
- MAC Address:** 02:04:14:02:30:0a
- IP Address:** 192.168.1.123
- Subnet Mask:** 255.255.254.0
- Gateway:** 192.168.1.1
- Primary DNS:** 192.168.1.1
- Secondary DNS:**
- Type:** Static IP Address

Figure 4-4

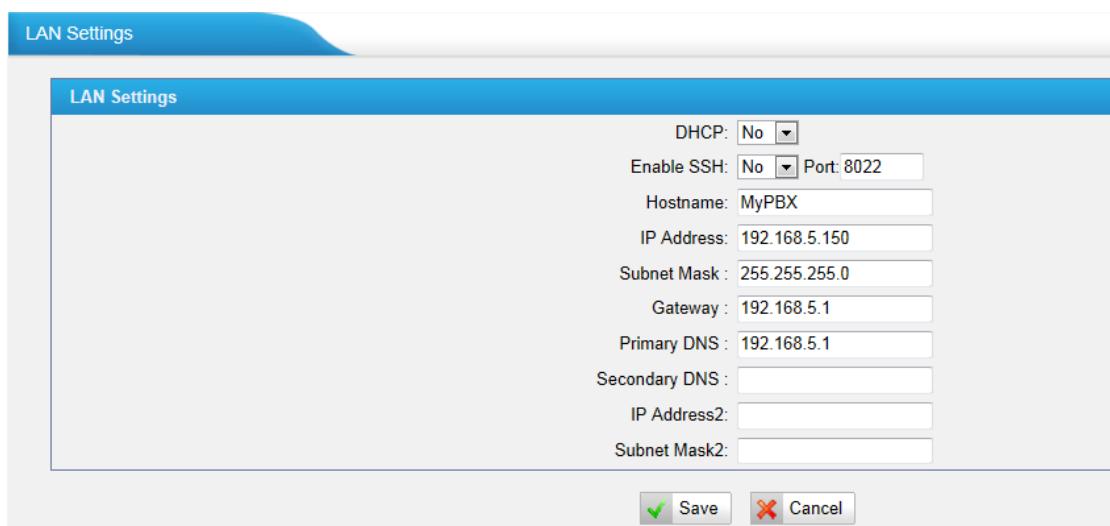
5 System

Click  to access.

In this page, we can configure the network settings, firewall settings, storage management and some other preferences like firmware update and hot standby.

5.1 Network Preferences

5.1.1 LAN Settings



The screenshot shows the 'LAN Settings' configuration page. At the top, there is a blue header bar with the title 'LAN Settings'. Below the header, there is a main configuration area with various input fields and dropdown menus. The fields include:

- DHCP: A dropdown menu set to 'No'.
- Enable SSH: A dropdown menu set to 'No'.
- Port: A text input field containing '8022'.
- Hostname: A text input field containing 'MyPBX'.
- IP Address: A text input field containing '192.168.5.150'.
- Subnet Mask: A text input field containing '255.255.255.0'.
- Gateway: A text input field containing '192.168.5.1'.
- Primary DNS: A text input field containing '192.168.5.1'.
- Secondary DNS: An empty text input field.
- IP Address2: An empty text input field.
- Subnet Mask2: An empty text input field.

At the bottom right of the configuration area, there are two buttons: a green checkmark button labeled 'Save' and a red X button labeled 'Cancel'.

Figure 5-1

DHCP

If this option is set, MyPBX will use DHCP to get an available IP address from your local network. Not recommended or you cannot access MyPBX without the right IP address

Enable SSH

This is the advance way to access the device, you can use the putty software to access the device. In the SSH access, you can do more advance setting and debug. Disabled by default.

Port: the default is 8022, you change it to another one

Hostname

Set the host name for MyPBX.

·IP Address

Set the IP Address for MyPBX.

Recommend to configure a static IP address for MyPBX

·Subnet Mask

Set the subnet mask for MyPBX.

·Gateway

Set the gateway for MyPBX.

·Primary DNS

Set the primary DNS for MyPBX.

·Secondary DNS

Set the secondary DNS for MyPBX.

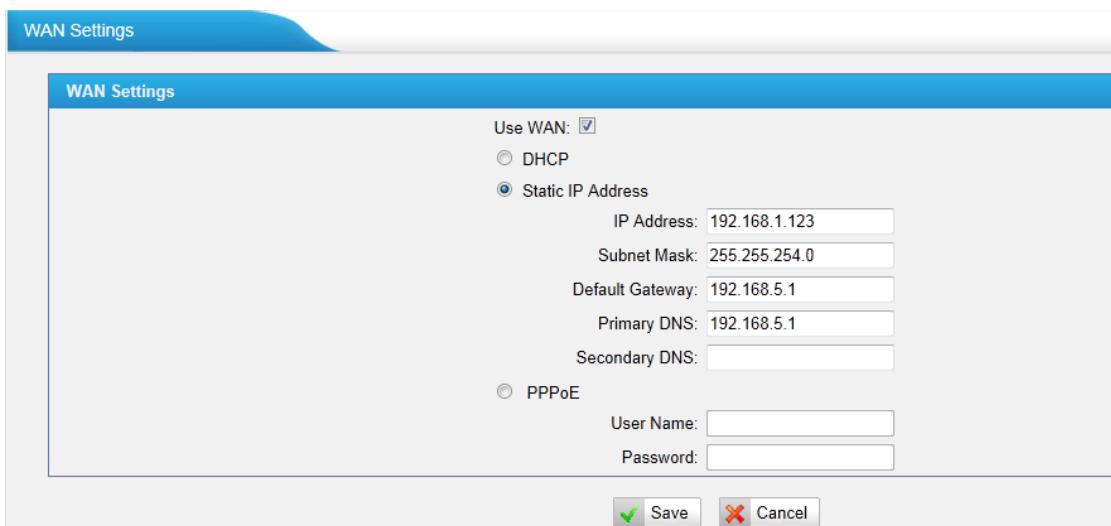
·IP Address2

Set the second IP Address for MyPBX.

·Subnet Mask2

Set the second subnet mask for MyPBX.

5.1.2 WAN Settings



The screenshot shows the 'WAN Settings' configuration page. At the top, there is a 'Use WAN:' checkbox which is checked. Below it, there are two radio buttons: 'DHCP' (unchecked) and 'Static IP Address' (checked). Under 'Static IP Address', there are four input fields: 'IP Address' (192.168.1.123), 'Subnet Mask' (255.255.254.0), 'Default Gateway' (192.168.5.1), and 'Primary DNS' (192.168.5.1). There is also a field for 'Secondary DNS' which is empty. Below these fields is a radio button for 'PPPoE', which is unchecked. Under 'PPPoE', there are three input fields: 'User Name' (empty), 'Password' (empty), and 'Save' and 'Cancel' buttons at the bottom right.

Figure 5-2

It support three connection types: DHCP (obtain an IP automatically), PPPoE, Static IP Address.

Note:

1. WAN port is disabled by default
2. WAN port cannot be used as a router to route the internet packages from WAN port to LAN port.

·DHCP

If your ISP says that you are connecting through DHCP or a dynamic IP address from your ISP, perform these steps:

- Step1: Select **DHCP** as the WAN Connection Type.
- Step2: Click **Save** button to save the settings.
- Step3: Reboot the device.
- Step4: Check the WAN's Status (Status→ Network status).

·Static IP Address

If your ISP says that you are connecting through a static or fixed IP address from your ISP, perform these steps:

- Step1: Select **Static IP Address** as the WAN Connection Type.
- Step2: Enter the IP Address.
- Step3: Enter the Subnet Mask.
- Step4: Enter the Gateway Address.
- Step5: Enter the Primary DNS and Secondary DNS.
- Step6: Click the **Save** button to save the settings.
- Step7: Reboot the device.
- Step8: Check the WAN's Status (Status→ Network status).

·PPPoE

If your DSL provider says that you are connecting through PPPoE or if you normally enter a user name and password to access the Internet, perform these steps:

- Step1: Select **PPPoE** as the WAN Connection Type.
- Step2: Enter the User Name.
- Step3: Enter the Password.
- Step4: Click the **Save** button to save the settings.
- Step5: Reboot the device.
- Step6: Check the WAN's Status (Status→ Network status)

5.1.3 DHCP Server

Dynamic Host Configuration Protocol (DHCP) is a network protocol that enables a server to automatically assign an IP address to a computer from a defined range of numbers (i.e., a scope) configured for a given network. You can set a local network NTP server for MyPBX here too

Note: MyPBX U300 can be working as a DHCP server, but cannot be regarded as an router. When using 'Phone Provisioning' for Grandstream IP phone, Enter the IP address of the server directly, e.g.:192.168.5.150; for other phones using the default configuration.

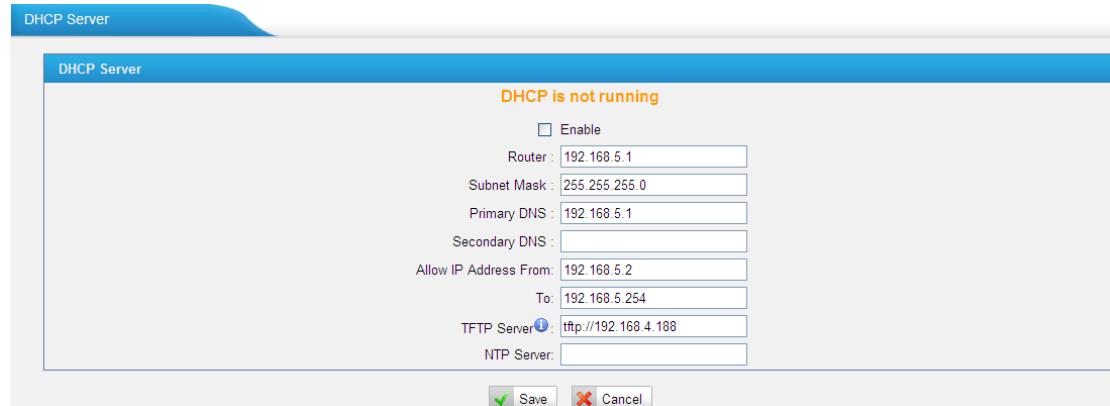


Figure 5-3

5.1.4 VLAN Settings

A VLAN(Virtual LAN) is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

Note:

MyPBX U300 is not the VLAN server, a 3-layer switch is still needed, please configure the VLAN information there first, then input the details in MyPBX, so that the packages via MyPBX will be added the VLAN label before sending to that switch.

VLAN Settings

Vlan Over Lan	
NO.1:	<input type="checkbox"/>
VLAN Number:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
VLAN Subnet Mask:	<input type="text"/>
Default Gateway:	<input type="text"/>
NO.2:	<input type="checkbox"/>
VLAN Number:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
VLAN Subnet Mask:	<input type="text"/>
Default Gateway:	<input type="text"/>
Vlan Over Wan	
NO.1:	<input type="checkbox"/>
VLAN Number:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
VLAN Subnet Mask:	<input type="text"/>
Default Gateway:	<input type="text"/>
NO.2:	<input type="checkbox"/>
VLAN Number:	<input type="text"/>
VLAN IP Address:	<input type="text"/>
VLAN Subnet Mask:	<input type="text"/>
Default Gateway:	<input type="text"/>
<input checked="" type="button"/> Save <input type="button"/> Cancel	

Figure 5-4

1) VLAN Over Lan

·NO.1

Click the NO.1 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Lan

·NO.2

Click the NO.2 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Lan.

2) VLAN Over Wan

·NO.1

Click the NO.1 you can edit the first VLAN over Wan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Wan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

·NO.2

Click the NO.2 you can edit the first VLAN over Wan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Wan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

5.1.5 VPN Settings

A virtual private network (VPN) is a method of computer networking--typically using the public internet--that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. MyPBX supports OpenVPN.

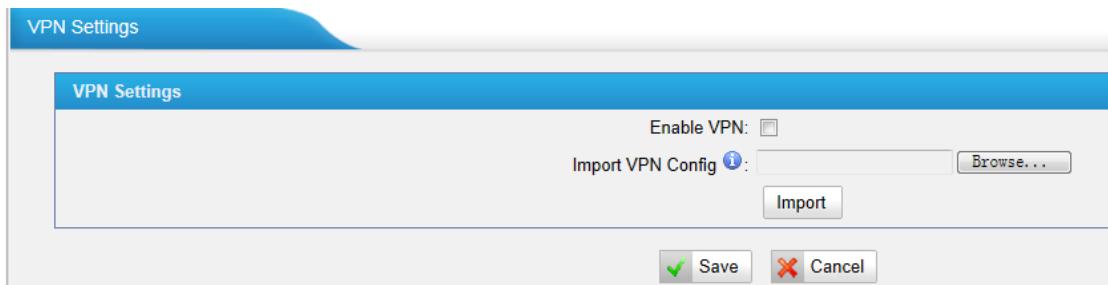


Figure 5-5

·Enable VPN

·Import VPN Config

Import configuration file of OpenVPN.

Note: Don't configure 'user' and 'group' in the 'config' file.

You can get the config package from the OpenVPN provider.

5.1.6 DDNS Settings

DDNS(Dynamic DNS) is a method / protocol / network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.

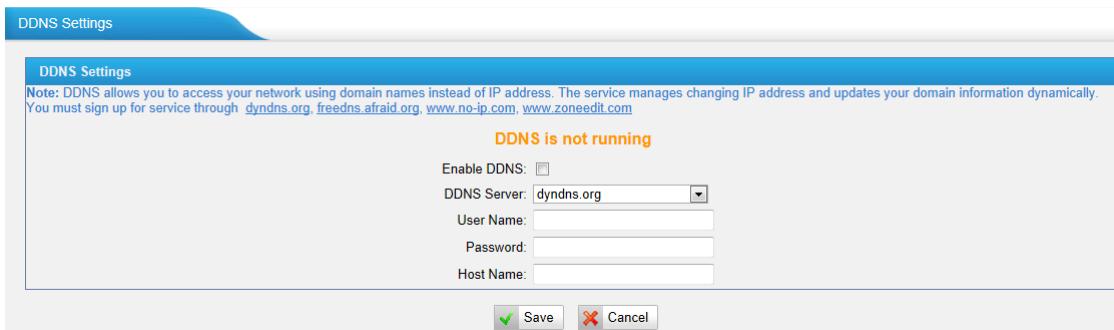


Figure 5-6

·Enable DDNS**·DDNS Server**

Select the DDNS server you sign up for service.

·User Name

User name the DDNS server provides you. This field is allowed to input 63 characters.

·Password

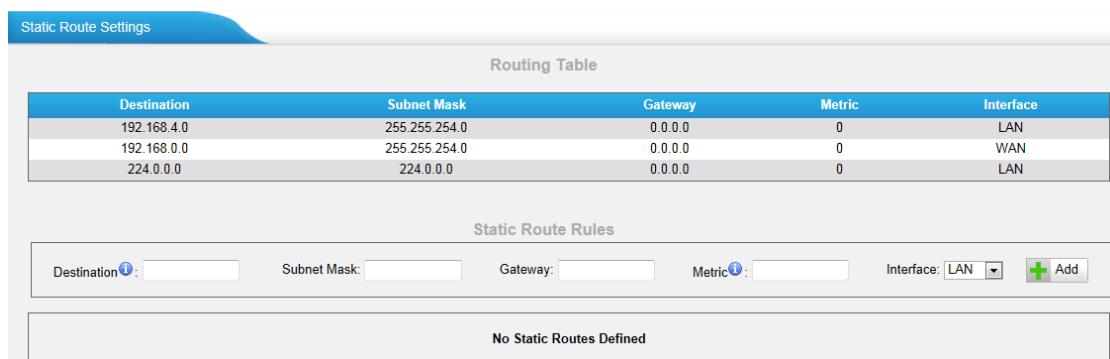
User account's password .

·Host Name

Note: DDNS allows you to access your network using domain names instead of IP address. The service manages changing IP address and updates your domain information dynamically. You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com

5.1.7 Static Route

MyPBX will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for MyPBX to force it goes out through different gateway when access to different internet. The default gateway priority of MyPBX from high to low is OpenVPN→WAN port→LAN port.



The screenshot shows the 'Static Route Settings' interface. It includes a 'Routing Table' section with the following data:

Destination	Subnet Mask	Gateway	Metric	Interface
192.168.4.0	255.255.254.0	0.0.0.0	0	LAN
192.168.0.0	255.255.254.0	0.0.0.0	0	WAN
224.0.0.0	224.0.0.0	0.0.0.0	0	LAN

Below the table is a 'Static Route Rules' section with fields for Destination, Subnet Mask, Gateway, Metric, and Interface, along with an 'Add' button. A message at the bottom states 'No Static Routes Defined'.

Figure 5-7

1) Route table

The current route rules of MyPBX

·Destination

The destination network to be accessed to by MyPBX

·Subnet Mask

Specify the destination network portion.

·Gateway

Define which gateway MyPBX will go through when access to the destination network.

·Metric

The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.

·Interface

Define which internet port to go through.

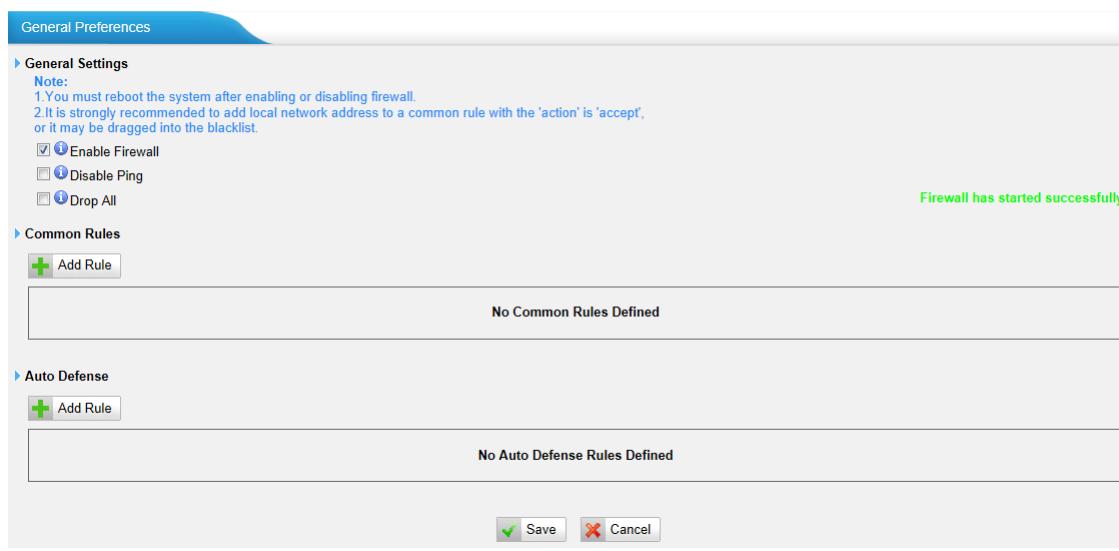
2) Static Route Rules

You can add new static route rules here.

5.2 Firewall Settings

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

5.2.1 Firewall Rules



The screenshot displays the 'General Preferences' section of the MyPBX U300 Administrator's Guide. It includes the following components:

- General Settings:** Contains checkboxes for 'Enable Firewall' (checked), 'Disable Ping' (unchecked), and 'Drop All' (unchecked). A note states: 'Note: 1.You must reboot the system after enabling or disabling firewall. 2.It is strongly recommended to add local network address to a common rule with the 'action' is 'accept', or it may be dragged into the blacklist.' A green status message 'Firewall has started successfully' is displayed.
- Common Rules:** Features an 'Add Rule' button and a list area showing 'No Common Rules Defined'.
- Auto Defense:** Features an 'Add Rule' button and a list area showing 'No Auto Defense Rules Defined'.
- Bottom Toolbar:** Includes 'Save' and 'Cancel' buttons.

Figure 5-8

1) General Settings

·Enable Firewall

Enable the firewall to protect the device. You should reboot the device to make the firewall run successfully.

·Disable Ping

Enable this item, net ping from remote hosts will be dropped.

·Drop All

When you enable 'Drop All' feature, system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one 'TCP' accept common rule must be created for port used for SSH access, port used for HTTP access and port used for CGI access.

2) Common Rules

There is not default rules inside, you can create them as required

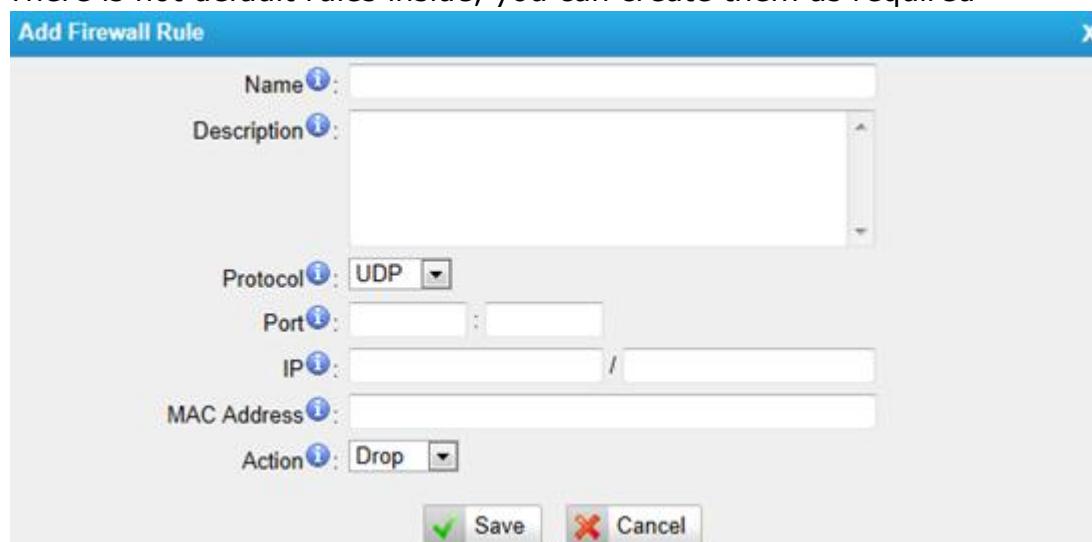


Figure 5-9

·Name

A name for this rule, e.g. 'HTTP'.

·Description

Simple description for this rule. E.g.: Accept the specific host to access the web interface for configuration.

·Protocol

The protocols for this rule.

·Port

Initial port should be on the left and end port should be on the right. The end port must be equal to or greater than start port.

·IP

The IP address for this rule. The format of IP address is: IP/mask

Ex:192.168.5.100/255.255.255.255 for IP 192.168.5.100

Ex:216.207.245.47/255.255.255.255 for IP 216.207.245.47

Ex:192.168.5.0/255.255.255.0 for IP from 192.168.5.0 to 192.168.5.255 .

·MAC Address

The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.

Note: The MAC address will be changed when it's remotely device, so it will not be working to filter using MAC for remote devices.

·Action

Accept: Accept the access from remote hosts.

Drop: Drop the access from remote hosts.

Ignore: Ignore the access.

5.2.2 IP blacklist

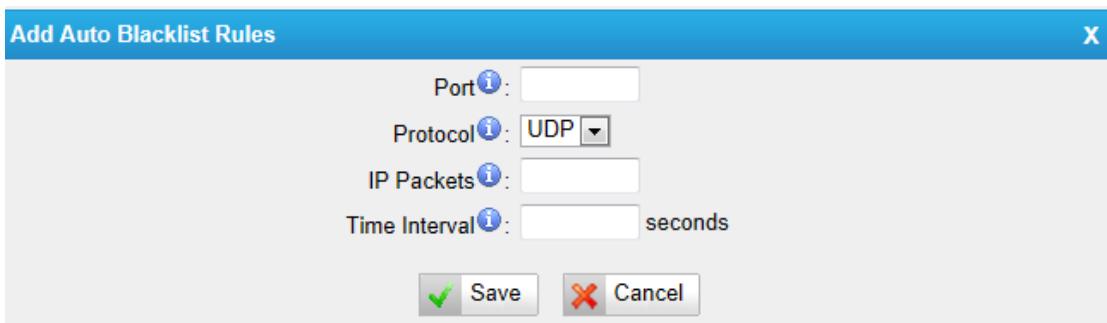
You can set some packets accept speed rules here. When a IP address which hasn't been accepted in common rules sends packets faster than the allowed speed, it will be set as black IP address and blocked automatically.



Figure 5-10

1) Blacklist rules

We can add the rules for IP blacklist rate as your demand



The dialog box has the following fields:

- Port: [Input field]
- Protocol: [Dropdown menu set to UDP]
- IP Packets: [Input field]
- Time Interval: [Input field] seconds
- Buttons: Save (green checkmark icon) and Cancel (red X icon)

Figure 5-11

•Port

Auto defense port

•Protocol

Auto defense protocol. TCP or UDP.

•IP Packets

Allowed IP packets number in the specific time interval.

•Time interval

The time interval to receive IP packets. For example, IP packets 90,time interval 60 means 90 IP packets are allowed in 60 seconds.

2) IP blacklist

The blocked IP address will display here, you can edit or delete it as your wish.

5.3 System Preference

In this page, we can set other system preference, like the password for admin account, system date and time, firmware update, hot standby, backup and restore, reset and reboot.

5.3.1 Password Settings

1) Change Password

•User

MyPBX support 3 levels users, choose one of them to change its password.

Administrator: have all authority

User Administrator: have basic authority; without the advanced authority to reset, update, backup and restore MyPBX.

CDR Administrator: only have the authority to check the call recordings.

•Enter Old Password

The default password of all users is '**password**'.

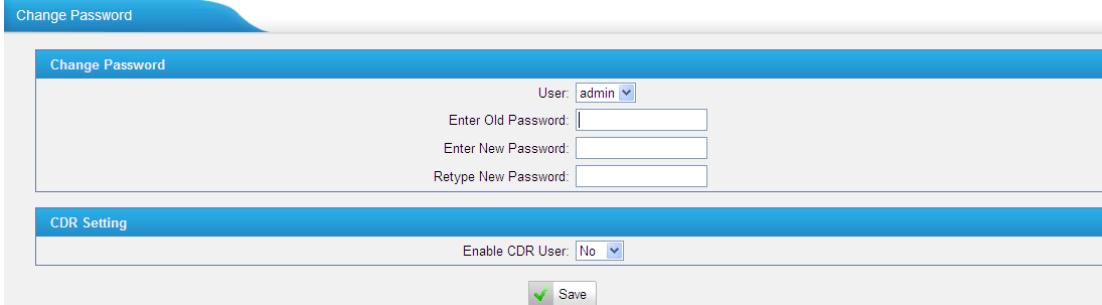
•Enter New Password**•Retype New Password**

To change the password, enter the new password and click update. The system will then prompt you re-login using your new password.

2) CDR settings

Whether enable CDR User.

Note: To logon MyPBX with CDR Administrator, enable CDR user first.

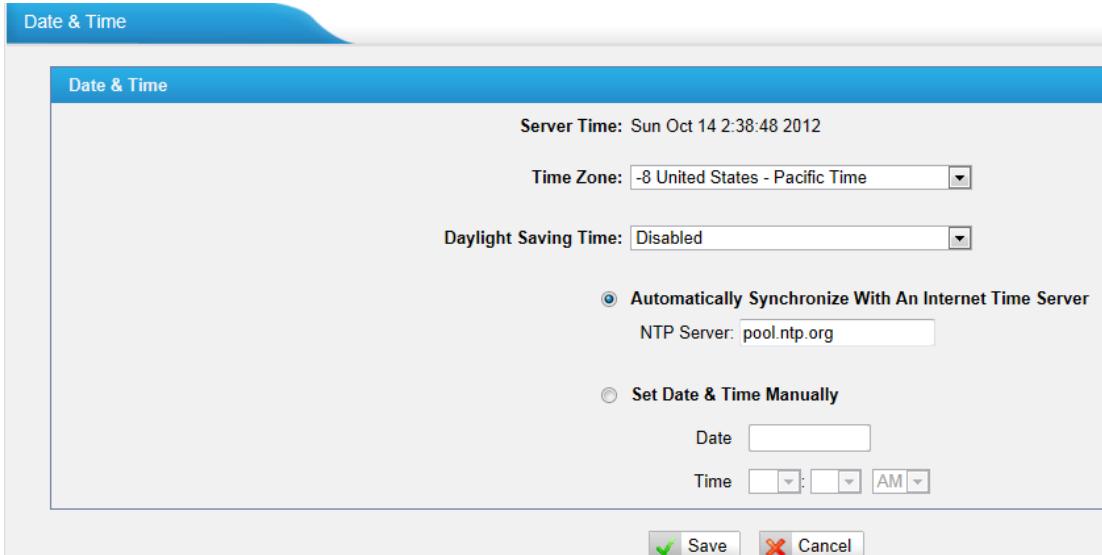


The screenshot shows two stacked configuration sections. The top section is titled 'Change Password' and contains fields for 'User' (set to 'admin'), 'Enter Old Password', 'Enter New Password', and 'Retype New Password'. The bottom section is titled 'CDR Setting' and contains a dropdown menu for 'Enable CDR User' (set to 'No') and a 'Save' button.

Figure 5-12

5.3.2 Date and Time

Set the date and time for MyPBX.



The screenshot shows the 'Date & Time' configuration page. It displays the 'Server Time' as 'Sun Oct 14 2:38:48 2012'. Below it are dropdown menus for 'Time Zone' ('-8 United States - Pacific Time') and 'Daylight Saving Time' ('Disabled'). There are two radio button options: one selected for 'Automatically Synchronize With An Internet Time Server' (with 'NTP Server' set to 'pool.ntp.org') and one for 'Set Date & Time Manually' (with fields for 'Date' and 'Time'). At the bottom are 'Save' and 'Cancel' buttons.

Figure 5-13

•Time Zone

You can choose your time zone here.

•Daylight Saving Time

Set the mode to Automatic or disabled

•Automatically Synchronize With an Internet Time Server

Input the NTP server so that MyPBX will update the time automatically

•Set Date & Time Manually

You can set the time to your local right time manually here

5.3.3 Firmware Update

Upgrading of the firmware is possible through the Administrator web interface using a TFTP Server or an HTTP URL.

Enter your TFTP Server IP address and firmware file location, then click start to update the firmware

Note:

1. If enabled 'Reset configuration to Factory Defaults', System will restore to factory default settings.
2. When update the firmware, please don't turn off the power. Or the system will get damaged.
3. More information for the steps to update the firmware, please refer to this link:
http://www.yeastar.com/download/MyPBX_U300_FirmwareUpgrade_en.pdf

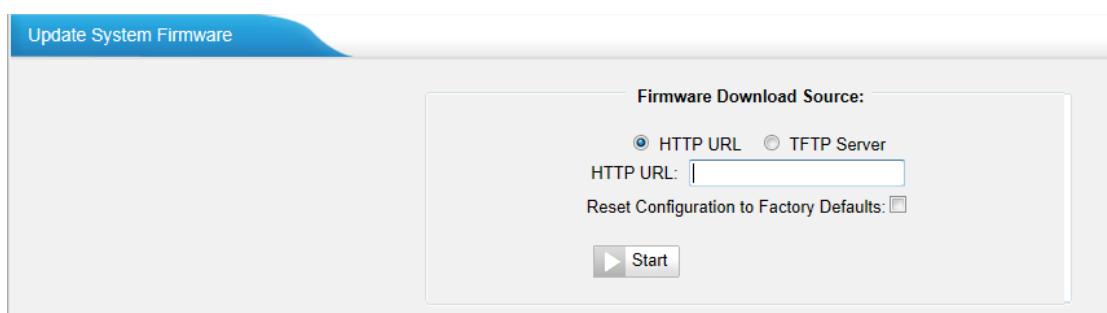


Figure 5-14

5.3.4 Hot Standby

Redundancy is achieved by using duplicate hardware and software installations and synchronizing data and operating state. Redundancy assures smooth operation even if a system goes down. Essentially a duplicate backup system takes over with virtual no loss of service. This technique assures absolute reliability no matter what failure occurs. In mission critical installations, redundancy is a way to address possibility of any failure.

Note 1: Before enabling the Host Standby feature, please make sure that the two servers in the failover pair are the same model, have the same modules installed in the same slots, the same hardware configurations and firmware version.

Note 2: Please configure the primary server first and configure the secondary server only after the running status of primary server becomes "active".

Note 3: The virtual IP address inputted in this page will be the one used for registering in each IP phone.

Note 4: Before configure the Email list in this page, please configure the 'voicemail settings' in "PBX→Basic settings", make sure the SMTP testing successfully.

Hot Standby

Note 1: Before enabling the Host Standby feature, please make sure that the two servers in the failover pair are the same model, have the same modules installed in the same slots, the same hardware configurations and firmware version.
Note 2: Please configure the primary server first and configure the secondary server only after the running status of primary server becomes "active".

Step 1: [Verify the basic information of the server](#)
Step 2: [Configure the IP address and hostname of the primary and secondary servers](#)
Step 3: [Configure Hot Standby Settings \(Example\)](#)

Basic
Running Status: Disabled Enable: No <input type="button" value=""/>
Mode Primary <input type="button" value=""/>
Secondary HostName <input type="text"/>
Secondary IP <input type="text"/>
Access Code <input type="text"/>
Virtual IP Address <input type="text"/>
Network Connection Detection <input type="text"/>
Down Notification
Notification Methods None <input type="button" value=""/>
Email List <input type="text"/>
Advanced <input type="button" value=""/>
Heartbeat Options
Keep Alive 2 <input type="text"/> s
Dead Time 120 <input type="text"/> s
<input checked="" type="button"/> Save <input type="button"/> Cancel

Figure 5-15

Mode: Primary means the main unit; Secondary means the standby unit;

Secondary/Primary Hostname: If this unit mode is primary, then you need to input the hostname of standby unit; vice versa, if this unit is selected as secondary, then the hostname of primary unit is required. In brief, you need to input each other's host name on this field.

IP: You need to input each other's IP address on this field.

Access code: To make an identification number to verify each other. The number must be the same to both units.

Virtual IP address: To fill in a virtual IP address includes mask, which is always points to the currently activated unit. Customer can register IP phones through this virtual IP address. Please make sure the virtual IP add includes mask is the same on both units but different from their former IP address.

Network Connection Detection: Generally it requires the IP address of the router or gateway that connects both units. MyPBX will connect another unit through this IP address.

Down Notification: The way of informing customer that the system down.

Keep Alive: Every 2 seconds, a package will be sent from one unit to another, which can test whether they are working properly.

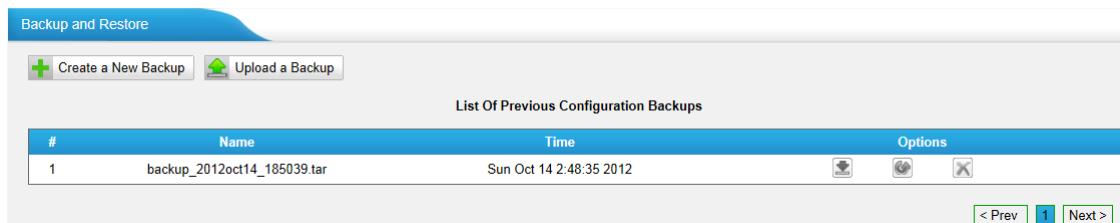
Dead Time: The default setting is 120 seconds. If there's no response within 120s after one receiving a package from the other, then the normal working unit will figure the other unit is dead and send an email to report the failure.

5.3.5 Backup and Restore

We can backup up the configurations before reset MyPBX U300 to factory defaults, and then restore it using this package.

Note:

1. Only configurations, custom prompts will be backed up, the voicemail and recording files are not included.
2. When you have update the firmware version, it's not recommend to restore using old package.

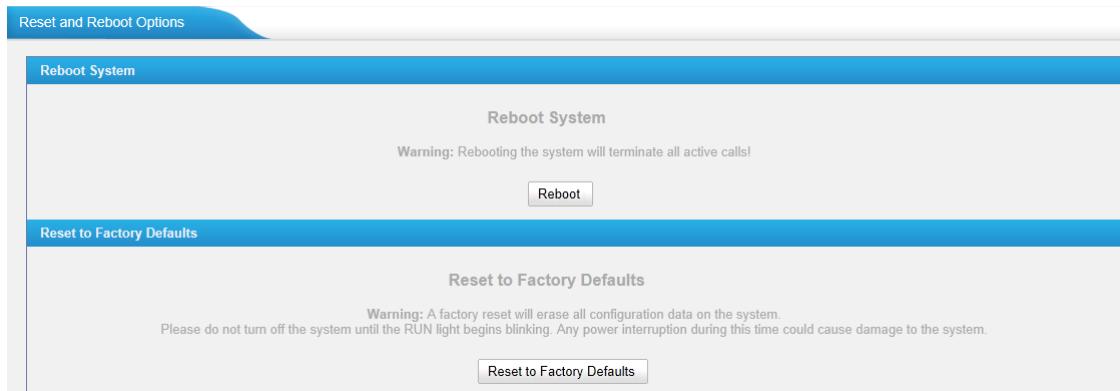


The screenshot shows a web-based interface titled 'Backup and Restore'. At the top, there are two buttons: 'Create a New Backup' (with a plus sign icon) and 'Upload a Backup' (with a cloud icon). Below these buttons is a table titled 'List Of Previous Configuration Backups'. The table has columns for '#', 'Name', 'Time', and 'Options'. There is one entry in the table: '# 1 Name backup_2012oct14_185039.tar Time Sun Oct 14 2:48:35 2012 Options'. At the bottom right of the table, there are navigation buttons: '< Prev', '1' (highlighted in blue), and 'Next >'.

Figure 5-16

5.3.6 Reset and Reboot

We can reset or reboot MyPBX U300 via web directly in this page .



The screenshot shows a web-based interface titled 'Reset and Reboot Options'. It contains two main sections: 'Reboot System' and 'Reset to Factory Defaults'.
Reboot System: This section has a button labeled 'Reboot' and a warning message: 'Warning: Rebooting the system will terminate all active calls!'.
Reset to Factory Defaults: This section has a button labeled 'Reset to Factory Defaults' and a warning message: 'Warning: A factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.'.

Figure 5-17

•Reboot System

Warning: Rebooting the system will terminate all active calls!

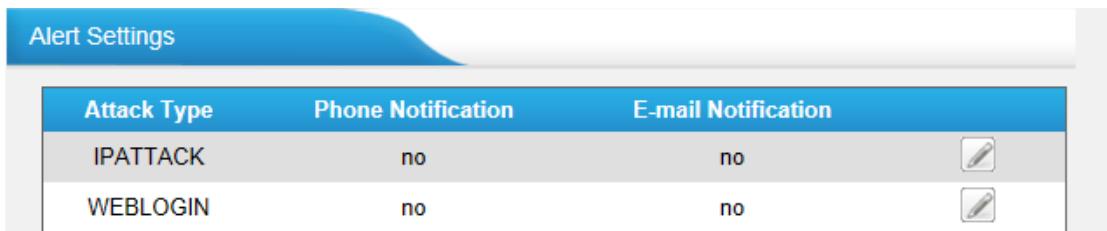
•Reset to Factory Defaults

Warning: A factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.

5.3.7 Alert Settings

If the device is attacked, the system will notify users the alert via call or E-mail. The attack mode includes IP attack and Web Login.

More details for the system security configuration please refer to **APPENDIX G MyPBX Security Configuration Guide**.



Attack Type	Phone Notification	E-mail Notification	
IPATTACK	no	no	
WEBLOGIN	no	no	

Figure 5.3.7.1

IPATTACK

When the system is attacked by an IP address, the firewall will add the IP to auto IP Blacklist and notify the user if it matches the protection rule.

Note: Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using Emails alert.

1) Phone Notification Settings

·Phone Notification:

Enable or disable phone notification

·Number

The numbers to call sending alert notification. Users can setup multiple extensions and outbound phone numbers. Please separate them by ';'. e.g. '500;81234'. If the extension has configured Follow Me Settings, the call would go to the forwarded number directly.

·Attempts

The attempts to dial the Number when there is no answer.

·Interval

The interval between each attempt to dial the phone number. Must be greater than 3 seconds

·Prompts

2) E-mail Notification Settings

·E-mail Notification

Enable or disable email notification

·To

Multiple mail address are allowed; please separate them by ';', e.g. 123@gmail.com;456@hotmail.com.

·Subject

The subject of email

·Content

Text content, support for predefined variables, Variable names and corresponding instructions are as follows:

`$(HOSTNAME)` host name

`$(LOCALIP)` local IP address

`$(SOURCEIP)` attack source IP address

`$(DESTMAC)` attacks destination mac (IPATTACK effective)

`$(DESTPORT)` attacks destination port number (IPATTACK effective)

`$(PROTOCOL)` protocol type (IPATTACK effective)

`$(DATETIME)` occurred

`$(USERNAME)` user name (WEBLOGIN effective)

`$(INTERFACE)` network interface name (IPATTACK effective)

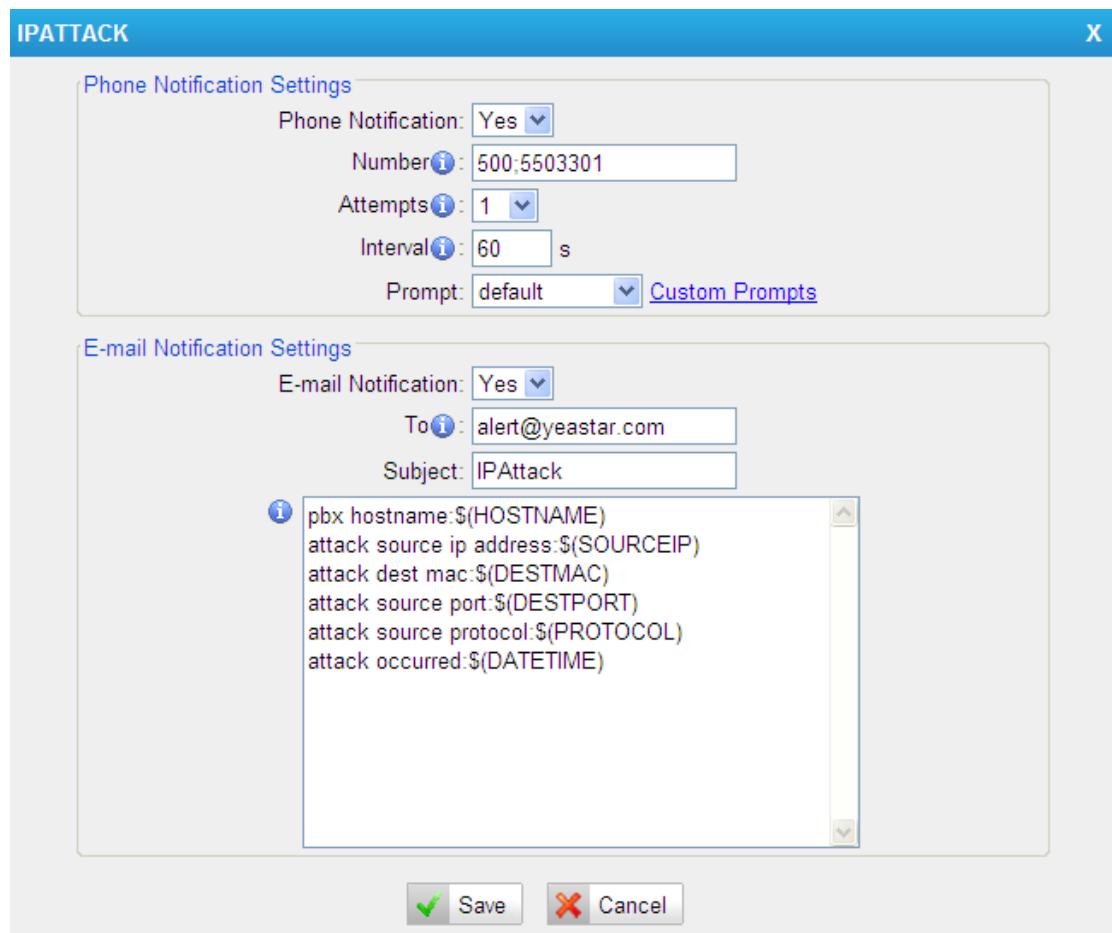


Figure 5.3.7.2

WEBLOGIN

Enter the password incorrectly five times to login MyPBX interface will be regarded as attack. The system will limit the IP login within 10 minutes and notify the user.

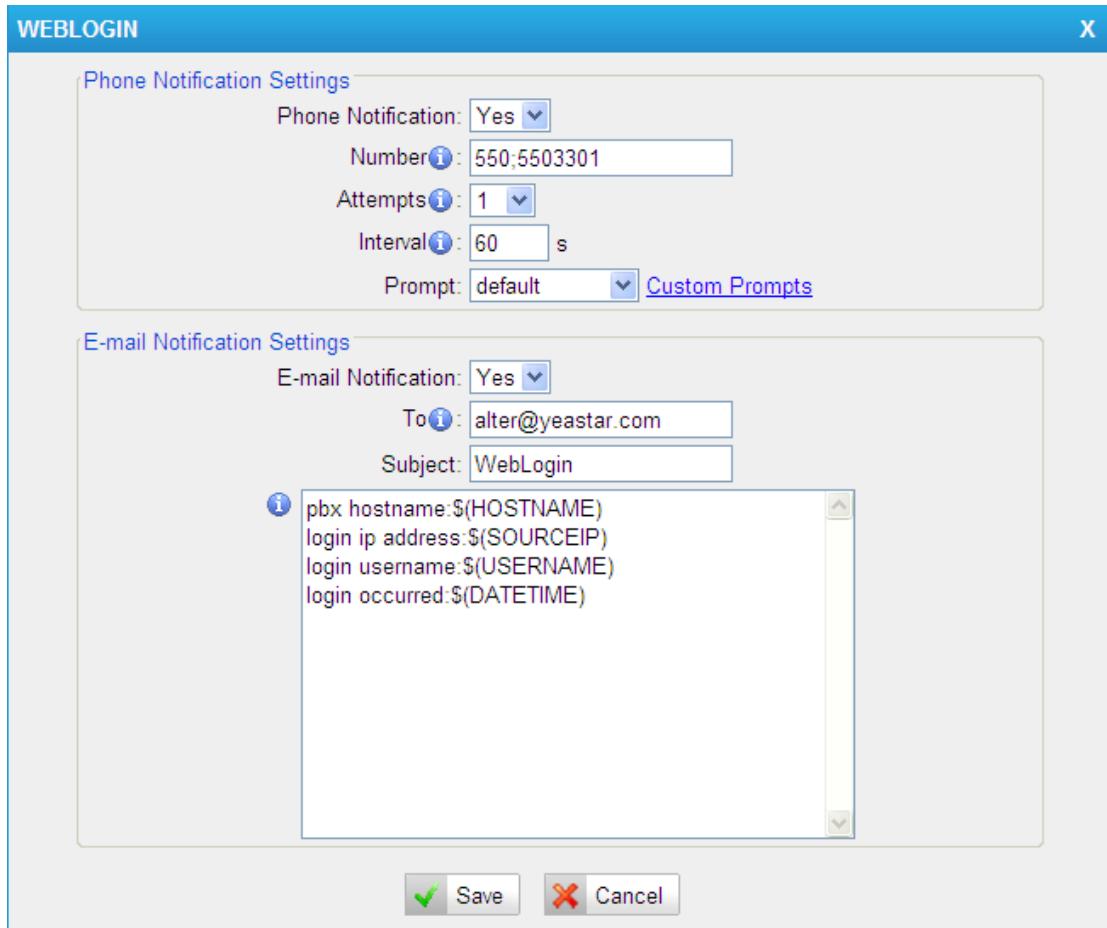


Figure 5.3.7.3

5.3.8 Addons

Call Recording enables all your inbound and outbound phone calls to be recorded. The system would detect any call conversation and record them automatically. The call recordings would be saved in the folder you specify and stored in the USB.

And if users want to use the function call recording, they need active this function firstly.

Note: For usage of the addon feature, please purchase activation code from your equipment provider.

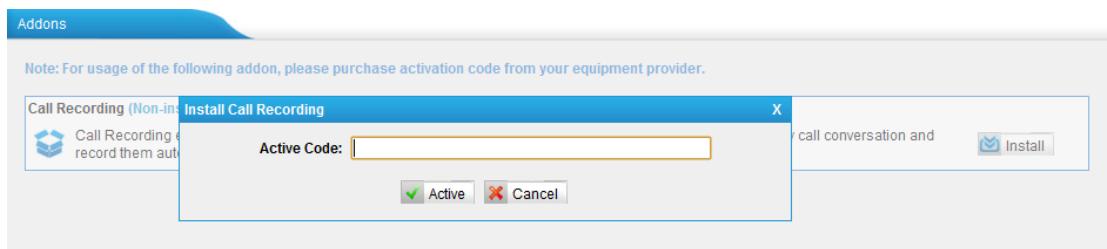


Figure 2.3.8.1

After activation, the page will show as below:

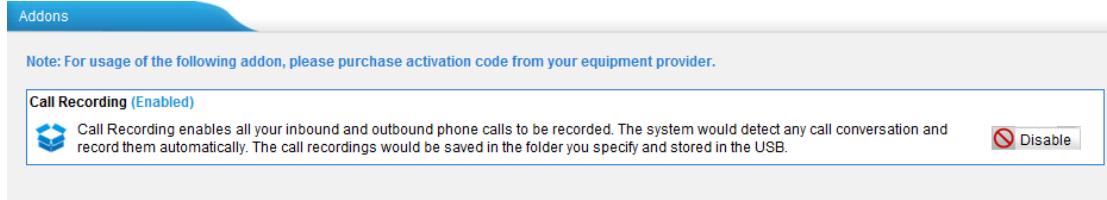


Figure 2.3.8.2

5.4 Storage Management

5.4.1 External Storage

The External Storage feature is used to extend storage space. Once configured, the files (voicemail, call recording files) created before the configured days will be moved to the Net-Disk.

Note: The shared folder must be based on Windows operation system.

Before external storage can be properly configured, an SMB share folder accessible from MyPBX must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

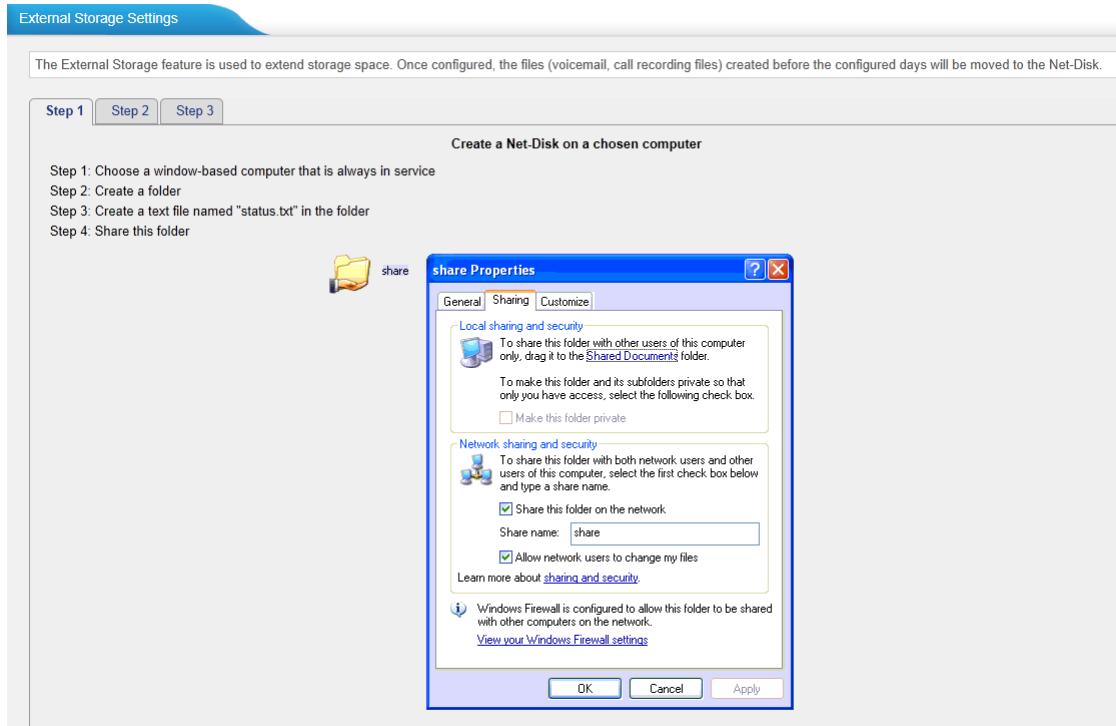


Figure 5-18

Step 1: Choose a window-based computer that is always in service

Step 2: Create a folder

Step 3: Create a text file named "status.txt" in the folder

Step 4: Share this folder

Then we need input the Net-Disk information in step2 page.

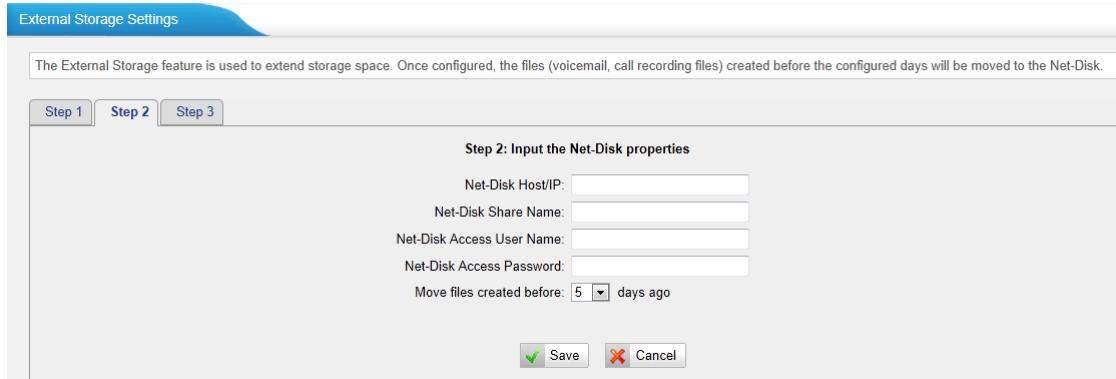


Figure 5-19

Net-Disk Host/IP: Change this to the IP address of the computer where backup files will be stored.

Net-Disk Share Name: Change this to the name of the shared folder where backups will be stored.

Net-DiskShare Username: The user name used to log into the network share. Leave this blank if it is not required

Net-DiskShare Password: The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders have been created. If the contents of the backup folder look similar to step3 page, then you have successfully configured external storage on the MyPBX unit.

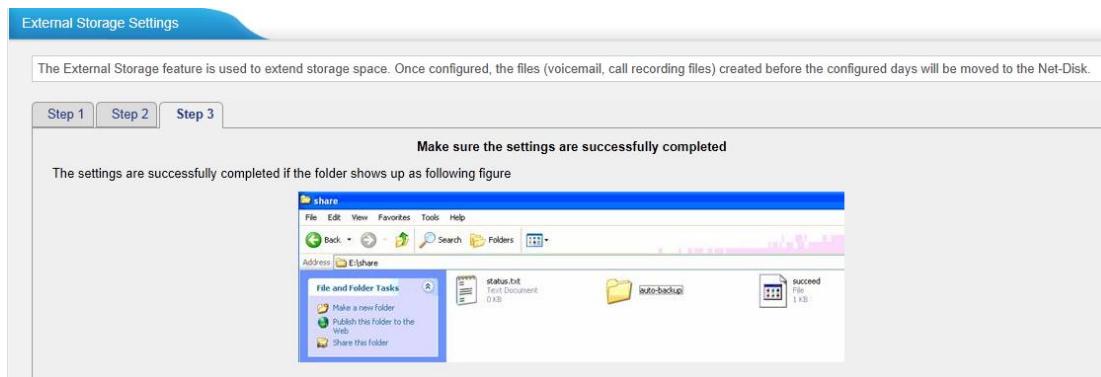


Figure 5-20

6 PBX

Click  to access

In this page, we can configure the settings of extension, trunk, inbound call control, outbound call control, audio settings and the others. When configured well, we can make calls as scheduled

6.1 Extensions

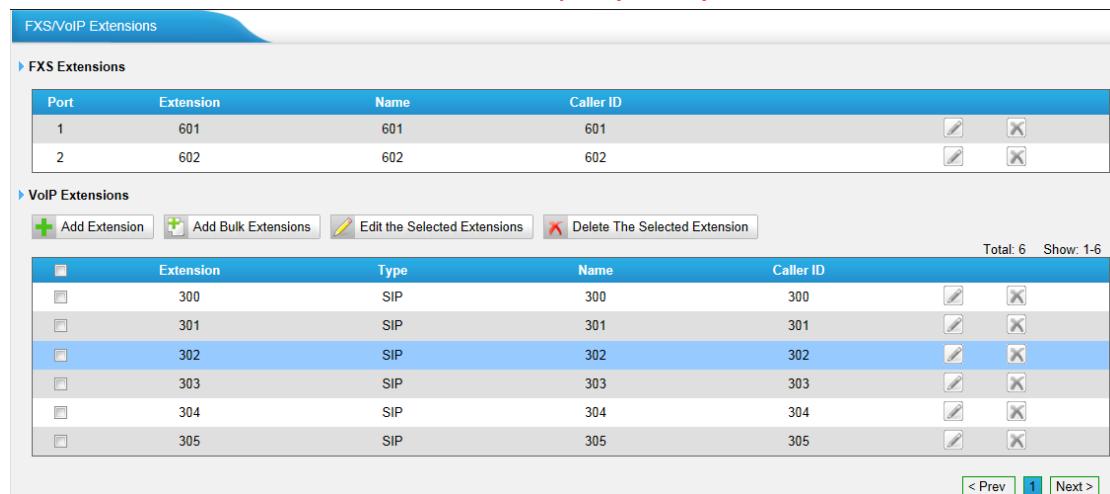
In this page, we can configure the extensions' details and provision the supported models automatically.

6.1.1 FXS/VoIP Extensions

There are three types of extensions supported in MyPBX U300: SIP, IAX and analog extension.

Note:

1. The number of SIP/IAX extension is 100
2. The number of FXS extension is only 2 (fixed)



The screenshot shows the 'FXS/VoIP Extensions' configuration interface. It is divided into two main sections: 'FXS Extensions' and 'VoIP Extensions'.

FXS Extensions: A table with columns: Port, Extension, Name, and Caller ID. It contains two entries:

Port	Extension	Name	Caller ID
1	601	601	601
2	602	602	602

VoIP Extensions: A table with columns: Extension, Type, Name, and Caller ID. It contains six entries, all of which are of type SIP:

Extension	Type	Name	Caller ID
300	SIP	300	300
301	SIP	301	301
302	SIP	302	302
303	SIP	303	303
304	SIP	304	304
305	SIP	305	305

Figure 6-1

FXS Extensions

FXS Extensions			
Port	Extension	Name	Caller ID
1	601	601	601
2	602	602	602

Figure 6-2

There are two fixed analog extensions in MyPBX U300, to modify the extension number, please delete it first, then recreate it again

1) General

Edit Extension - 601

General	Other Settings				
General <table border="1"> <tr> <td>Extension <input type="text" value="601"/></td> <td>Port: <input type="text" value="1"/></td> </tr> <tr> <td>Name <input type="text" value="601"/></td> <td>Caller ID <input type="text" value="601"/></td> </tr> </table>		Extension <input type="text" value="601"/>	Port: <input type="text" value="1"/>	Name <input type="text" value="601"/>	Caller ID <input type="text" value="601"/>
Extension <input type="text" value="601"/>	Port: <input type="text" value="1"/>				
Name <input type="text" value="601"/>	Caller ID <input type="text" value="601"/>				
Voicemail <table border="1"> <tr> <td><input checked="" type="checkbox"/> Enable Voicemail <input type="text" value="601"/></td> <td>Voicemail Access PIN # <input type="text" value="601"/></td> </tr> </table>		<input checked="" type="checkbox"/> Enable Voicemail <input type="text" value="601"/>	Voicemail Access PIN # <input type="text" value="601"/>		
<input checked="" type="checkbox"/> Enable Voicemail <input type="text" value="601"/>	Voicemail Access PIN # <input type="text" value="601"/>				
Mail Setting <table border="1"> <tr> <td><input type="checkbox"/> Enable Send Voicemail <input type="text" value="1000"/></td> <td>ms</td> </tr> <tr> <td>Email Address <input type="text"/></td> <td></td> </tr> </table> <p>Note: Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.</p>		<input type="checkbox"/> Enable Send Voicemail <input type="text" value="1000"/>	ms	Email Address <input type="text"/>	
<input type="checkbox"/> Enable Send Voicemail <input type="text" value="1000"/>	ms				
Email Address <input type="text"/>					
Flash <table border="1"> <tr> <td>Hook Flash Detection <input type="text" value="1000"/></td> <td>ms</td> </tr> </table>		Hook Flash Detection <input type="text" value="1000"/>	ms		
Hook Flash Detection <input type="text" value="1000"/>	ms				
Group <table border="1"> <tr> <td>Pickup Group <input type="text" value="--"/></td> </tr> </table>		Pickup Group <input type="text" value="--"/>			
Pickup Group <input type="text" value="--"/>					
Call Duration Setting <table border="1"> <tr> <td>Max Call Duration <input type="text"/></td> <td>s</td> </tr> </table>		Max Call Duration <input type="text"/>	s		
Max Call Duration <input type="text"/>	s				
<input type="button" value="Save"/> <input type="button" value="Cancel"/>					

Figure 6-3

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Port

The extension correspond port.

·Name

A character-based name for this user, i.e. 'Bob Jones' .

·Caller ID

The Caller ID (CID) string will be used when this user calls another internal user.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #

Voicemail Password for this extension, i.e. '1234'.

3) Mail Setting

·Enable Send Voicemail

Once enabled, the voicemail will be sent to the belowemail address as an attachment.

·Send Voicemail to Email Address

This option defines whether or not voicemails/Fax is sent to the Email address asan attachment.

Note: Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

4) Flash

·Hook Flash Detection

Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MyPBX before it will recognize a second event. If a second event occurs in less time than defined byHook Flash Detection, then MyPBX will ignore the event. The default value of Flash is 1000 ms, and it can be configured in 1ms increments.

5) Group

·Pickup Group

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default *4).

Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

6) Call Duration Settings

Setup the max cull duration for every call of this extension, but it's only valid for outbound calls. And if enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

Note: This setting will not be valid for internal calls.

7) Other options

•Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the 'When busy' follow me options will not be available.

•DND

Don't Disturb.

•User Web Interface

Check this option to allow the user to login to the MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

•Ring Out

Check this option if you want to custom the ring time. Tone will stop over the time defined

8) Follow me (Call Forwarding)

This function sets inbound call forwarding on an extension. An administrator can configure Follow Me for this extension

9) Volume Settings

Rxgain: The Volume sent to FXS extension.

Txgain: The Volume sent out by the FXS extension

10) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (Your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension.

Note: If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled

11) Spy Settings

MyPBX allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

·spy modes

There are 4 spy modes available for choice:

General spy: you have the permission to use the following 3 modes.

Normal spy: you can only hear the call, but can't talk

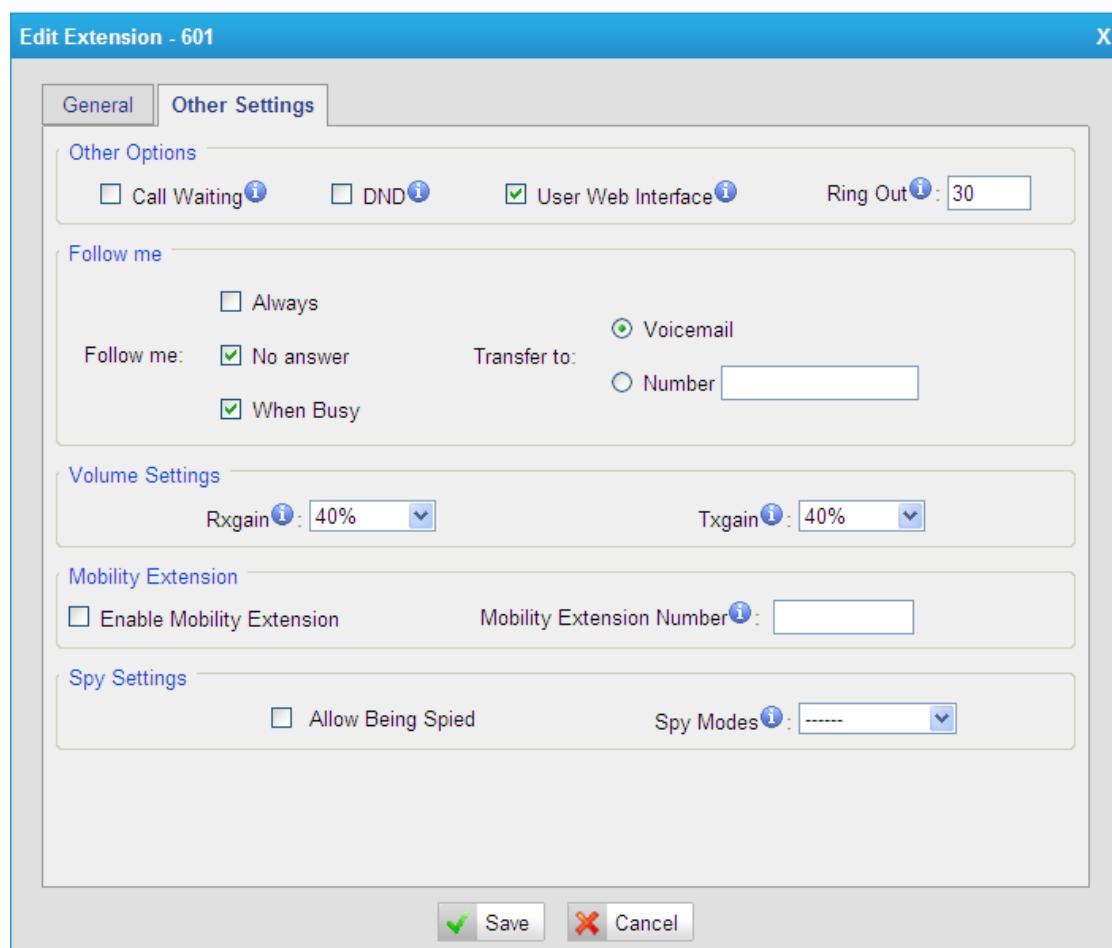
Whisper spy: you can hear the call, and can talk with the monitored extension

Barge spy: you can hear the call and talk with them both

Note: for example, if 500 want to monitor extension 501, we need to enable the 'allow being spied' for 501, and choose the spy mode for extension 500.

Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call.

If 500 choose 'normal spy', it should dial'*90501' to start monitoring;
if 500 choose 'whisper spy', it should dial '*91501' to start monitoring;
if 500 choose 'barge spy', it should dial '*92501' to start monitor;
if 500 choose 'general spy', it can dial '*90501','*91501' or '*92501' to start monitor.

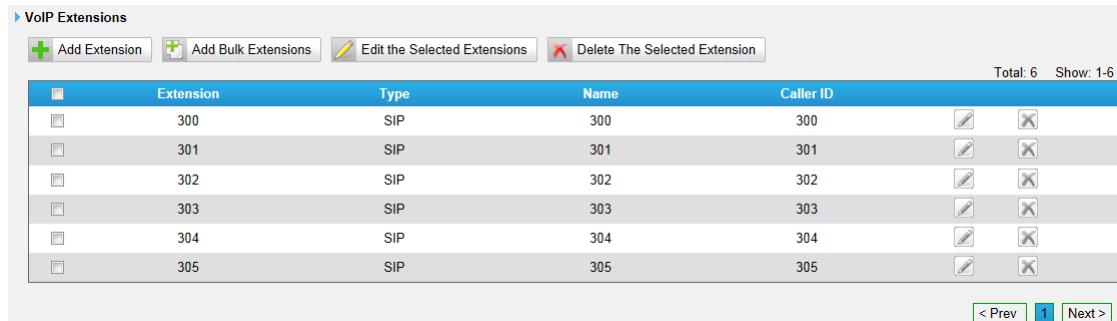


The screenshot shows the 'Edit Extension - 601' dialog box with the 'Other Settings' tab selected. The 'Spy Settings' section contains two main fields: a checked checkbox labeled 'Allow Being Spied' and a dropdown menu labeled 'Spy Modes' which is currently empty ('----').

Figure 6-4

VoIP Extensions

A VOIP extension is a SIP/IAX Account that allows an IP Phone or an IP Soft-Phone client to register on MyPBX



The screenshot shows a table titled 'VoIP Extensions' with the following data:

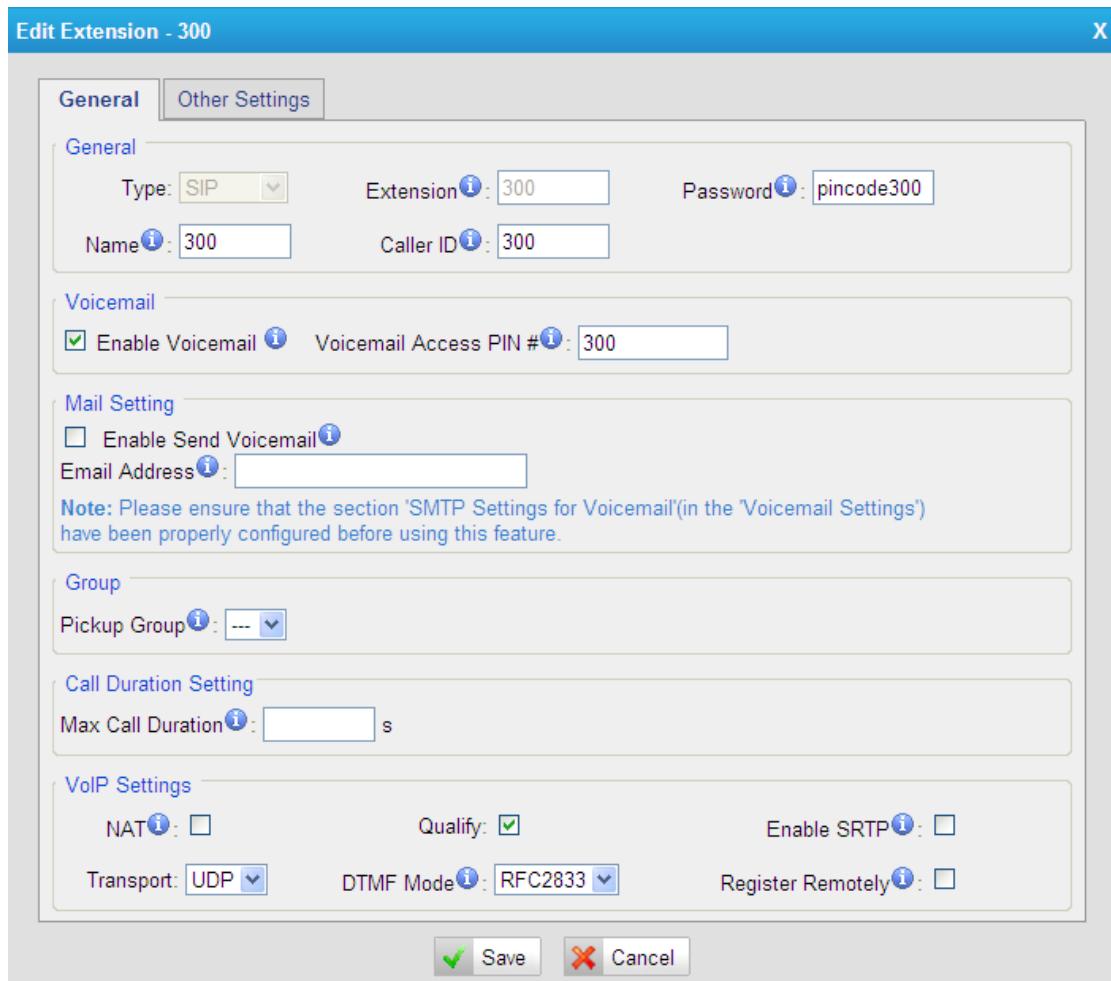
	Extension	Type	Name	Caller ID		
<input type="checkbox"/>	300	SIP	300	300		
<input type="checkbox"/>	301	SIP	301	301		
<input type="checkbox"/>	302	SIP	302	302		
<input type="checkbox"/>	303	SIP	303	303		
<input type="checkbox"/>	304	SIP	304	304		
<input type="checkbox"/>	305	SIP	305	305		

Total: 6 Show: 1-6

< Prev | 1 | Next >

Figure 6-5

We can click 'Add extension' to start.



Edit Extension - 300

General **Other Settings**

General

- Type: **SIP**
- Extension **i: 300**
- Password **i: pincode300**
- Name **i: 300**
- Caller ID **i: 300**

Voicemail

- Enable Voicemail **i: Voicemail Access PIN #i: 300**

Mail Setting

- Enable Send Voicemail **i: Email Address i:**

Note: Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

Group

- Pickup Group **i: ---**

Call Duration Setting

- Max Call Duration **i:** s

VoIP Settings

- NAT **i:**
- Qualify **i:**
- Enable SRTP **i:**
- Transport: **UDP**
- DTMF Mode **i: RFC2833**
- Register Remotely **i:**

Save **Cancel**

Figure 6-6

1) General

·Type

Extension type: SIP,IAX or SIP/IAX.

SIP – The extension sends and receives calls using the VoIP protocol SIP.

IAX -The extension sends and receives calls using the VoIP protocol IAX.

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Password

The password for this extension, Ex: '12t3f6'

·Name

A character-based name for this user, EX: 'Bob Jones'

·Caller ID

The Caller ID will be used when this user calls another internal extension.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #

The voicemail Password for this extension, i.e. '1234'.

3) Mail Setting

This option defines whether or not voicemails or faxes are sent to an Email Address as attachment.

·Enable Send Voicemail

Once enabled, the voicemail will be sent to email as an attachment.

·Email Address

Email address used to receive the voicemail or Fax.

Note: Please ensure that the section 'SMTP Settings For Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

4) Group

·Pickup Group

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default is*4).

Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

5) Call Duration Settings

Setup the max call duration for every call of this extension, but it's only valid for outbound calls. And if enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

Note: This setting will not be valid for internal calls.

6) VoIP Settings

NAT

This setting should be used when the system is using a public IP address to communicate with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.

Quality

Send check alive packets to IP phones

Enable SRTP

Enable extension for SRTP (RTP Encryption).

Transport

This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.

DTMF Mode – RFC2833, Info, Inband, Auto.

Remote Register

Allow to register remote extensions.

This option is used to enhance the system security, it's disabled by default.

More details for the system security configuration please refer to **APPENDIX G MyPBX Security Configuration Guide**.

7) Other Options

Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the 'When busy' follow me options will not be available. The call waiting function of IP phone has higher priority than MyPBX's call waiting function.

DND

Don't Disturb. When DND is enabled for an extension, the extension will be not available.

User Web Interface

Check this option to allow the user to login to the MyPBX User Web interface,

which can be used to check voicemail and extension recordings. Users may login to MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

.Ring Out

Check this option if you want to customize the ring time. Ring tone will stop over the time defined.

8) Follow me (Call Forwarding)

Call forwarding for an extension can be configured here. The administrator can configure Follow Me option for this extension. If you want to transfer the call to an outbound number, please follow the dial pattern of outbound route filled in the outbound number.

For example: transferring to your mobile phone number 123456789, the dial pattern of outbound route is '9.', you should fill in 9123456789 here.

9) IP Restriction

·Enable IP Restriction

Check this option to enhance the VoIP security for MyPBX. If this option is enabled, only the permitted IP/Subnet mask will be able to register this extension number. In this way, the VoIP security will be enhanced.

·Permitted 'IP address/Subnet mask'

The input format should be 'IP address'+'/'+'Subnet mask'.

e.g."192.168.5.100/255.255.255.255" means only the device whose IP address is 192.168.5.100 is allowed to register this extension number.

e.g."192.168.5.0/255.255.255.0" means only the device whose IP address is 192.168.5.XXX is allowed to register this extension number.

10) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (Your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension.

·Mobility Extension Number

Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number filed. E.g. if you want to set "15960XXXXXX" as mobile extension, and the dial pattern of the outbound route is "9"; you should set "915960XXXXXX" here.

Note: If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled

11) Spy Settings

MyPBX allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

·spy modes

There are 4 spy modes available for choice:

General spy: you have the permission to use the following 3 modes.

Normal spy: you can only hear the call, but can't talk

Whisper spy: you can hear the call, and can talk with the monitored extension

Barge spy: you can hear the call and talk with them both

Note: for example,if 500 want to monitor extension 501, we need to enable the 'allow being spied ' for 501, and choose the spy mode for extension 500.

Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call

If 500 choose 'normal spy', it should dial'*90501' to start monitoring;

If 500 choose 'whisper spy', it should dial '*91501' to start monitoring;

If 500 choose 'barge spy', it should dial '*92501' to start monitor;

If 500 choose 'general spy', it can dial '*90501','*91501' or '*92501' to start monitor.

Edit Extension - 300

<input checked="" type="radio"/> General	<input type="radio"/> Other Settings
Other Options <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Call Waiting <small>i</small> <input type="checkbox"/> DND <small>i</small> <input checked="" type="checkbox"/> User Web Interface <small>i</small> Ring Out <small>i</small>: <input type="text" value="30"/> </div>	
Follow me <div style="display: flex; justify-content: space-between;"> <div style="flex: 1;"> <input type="checkbox"/> Always Follow me: <input checked="" type="checkbox"/> No answer <input type="checkbox"/> When Busy </div> <div style="flex: 1;"> Transfer to: <div style="display: flex; align-items: center;"> <input checked="" type="radio"/> Voicemail <input type="radio"/> Number <input type="text"/> </div> </div> </div>	
IP Restriction <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Enable IP Restriction <small>i</small> </div> <div style="display: flex; justify-content: space-between;"> Permitted 'IP address/Subnet mask' 1 <small>i</small>: <input type="text"/> Permitted 'IP address/Subnet mask' 2 <small>i</small>: <input type="text"/> Permitted 'IP address/Subnet mask' 3 <small>i</small>: <input type="text"/> Permitted 'IP address/Subnet mask' 4 <small>i</small>: <input type="text"/> </div>	
Mobility Extension <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Enable Mobility Extension Mobility Extension Number <small>i</small>: <input type="text"/> </div>	
Spy Settings <div style="display: flex; justify-content: space-between;"> <input type="checkbox"/> Allow Being Spied Spy Modes <small>i</small>: <input style="width: 100px;" type="text"/> </div>	
<input checked="" type="button"/> Save <input type="button"/> Cancel	

Figure 6-7

6.1.2 Phone Provisioning

The Auto Provision sub menu provides users a method to Auto Provision IP Phone after the Express Setup process.

Note: Auto Provision functions fully test with these models:

Yealink (T12,T18,T20,T22,T26,T28,T32,T38,VP530,VP-2009)

Snom (300,320,360,370)

Polycom (IP 6000,IP 7000,IP 32X,IP33X,IP430,IP450,IP550,IP560,VVX1500)

Cisco (IP7940,IP7960)

Aastra(480i,480i CT,6757i,6757i CT, 6737i)

Grandstream(GXP1450,GXP2100,GXP2110,GXP2120)

Escene(ES220,ES320,ES330,ES410,ES620)

Fanvil(C56,C58,C60,C62)

News:

When provisioning Yealink and Snom IP phone, MyPBX is not needed to be set as the only DHCP server any more

Phone Provisioning

- ▶ General Settings for Yealink
- ▶ General Settings for Aastra
- ▶ Phone Book
- ▶ Configured Phone

Total: 0 Show: 0-0 View: 15

Mac Address List				
Not Configured Phone				
<input type="button" value="Configure the Selected Phones"/> <input type="button" value="Refresh"/>				
	ID	MAC Address	Manufacturer	Phone Type
<input type="checkbox"/>	1	001565113844	Yealink	--
<input type="checkbox"/>	2	001565114094	Yealink	--
<input type="checkbox"/>	3	0015651be4a4	Yealink	--
<input type="checkbox"/>	4	001565132139	Yealink	--
<input type="checkbox"/>	5	0015651be494	Yealink	--
<input type="checkbox"/>	6	0015652c2cc8	Yealink	--
<input type="checkbox"/>	7	00156511189e	Yealink	--
<input type="checkbox"/>	8	0015652c3f80	Yealink	--
<input type="checkbox"/>	9	0015653416c6	Yealink	--
<input type="checkbox"/>	10	001565116da8	Yealink	--
<input type="checkbox"/>	11	0015653411ea	Yealink	--
<input type="checkbox"/>	12	00156534168e	Yealink	--
<input type="checkbox"/>	13	0015651208b3	Yealink	--
<input type="checkbox"/>	14	001565111886	Yealink	--
<input type="checkbox"/>	15	001565298bac	Yealink	--

No Files Found.

Figure 6-8

6.1.2.1 General Settings for Yealink

In this page, you can configure it before provisioning Yealink IP phones, including the items like general preferences, codecs, remote phone book and firmware upgrade

Note: if firmware download server is enabled, IP phone will update the firmware automatically according the version and server you have configured during the provision process.

General Settings for Yealink

[Go Back to Phone Provisioning](#)

Language <small>i</small> : <input type="text" value="English"/>
Web server Type: <input type="text" value="HTTP&HTTPS"/>
Admin Password: <input type="radio"/> Fixed <input type="radio"/> Prefix <input type="text" value="admin"/>
Time Zone: <input type="text" value="+8 China(Beijing)"/>
Primary NTP Server: <input type="text" value="cn.pool.ntp.org"/>
Secondary NTP Server: <input type="text" value="cn.pool.ntp.org"/>
Daylight Saving Time: <input type="text" value="Disabled"/>
Time Format: <input type="text" value="12 Hour"/>
Date Format: <input type="text" value="WWW MMM DD"/>
Voicemail: <input type="text" value="Yes"/>
PNP URL: <input type="radio"/> Automatic <input type="radio"/> Custom

Figure 6-9

6.1.2.2 General Settings for Aastra

In this page, you can configure it before provisioning Aastra IP phones, including the items like local dial plan, time and date setting programkeys configuration and so on.

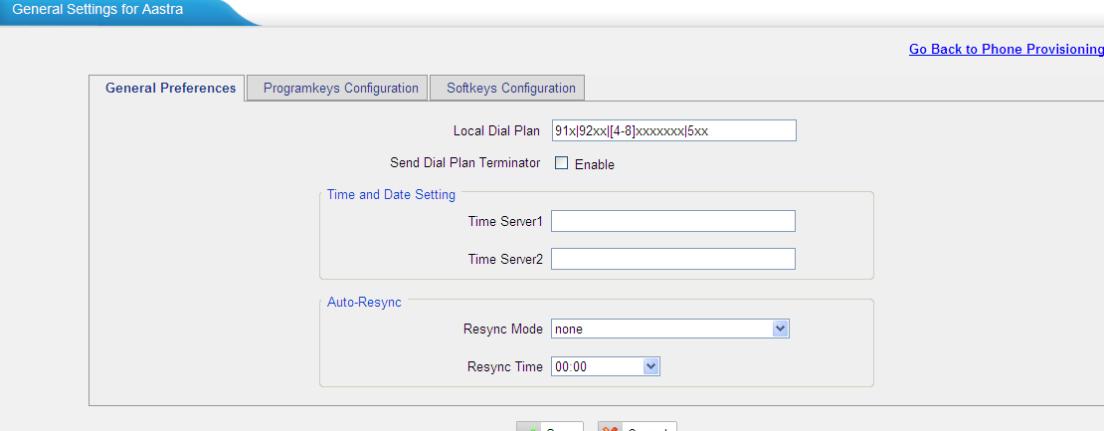


Figure 6-10

6.1.2.3 Phone book

You can add your contacts here and provision them to your IP phone

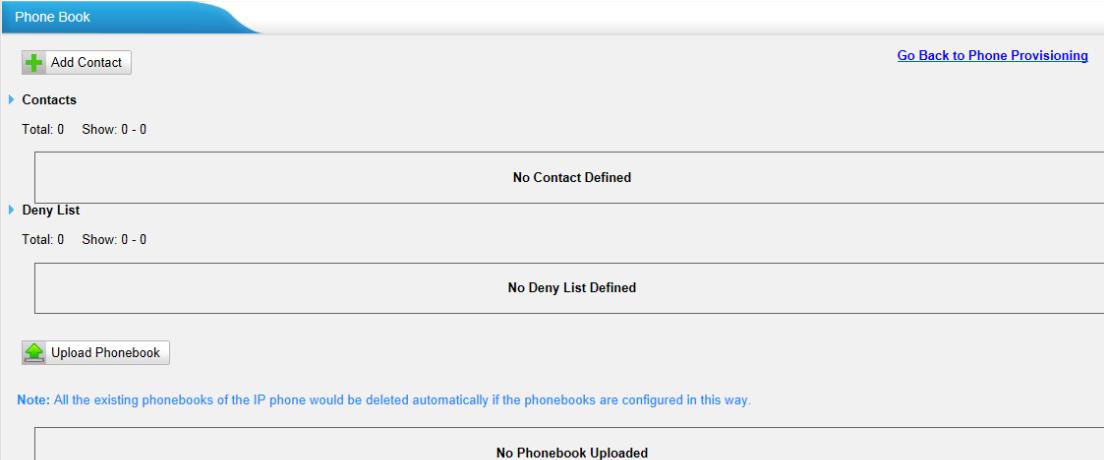


Figure 6-11

1) Add Contact

Type

There are three types: None, VIP and Deny list (Blacklist).

Group

There are 5 groups: None, Friends, Family, Work, Colleagues list.

Nick Name

You can set a nick name for this number.

Favorite

Only works with snom phone.

·Organization

Input the organization of this contact. Only works with snom phone.

·Title

Input the title of this contact. Only works with snom phone.

·Email

Input the email of this contact. Only works with snom phone.

·Birthday

Input the birthday of this contact. Only works with snom phone.

·First Name

Input the first name of this contact. Only works with snom phone.

·Family Name

Input the family of this contact. Only works with snom phone.

·Office Number

Input the office number here

·Mobile Number

Input the mobile number here

·Home Number

Input the home number here

·Sub Number

Add sub number of this contact. Only works with snom phone.

·Note

Take some note of this contact. Only works with snom phone.

Add Contact

Type: <input type="button" value="None"/>	Group: <input type="button" value="None"/>
Nick Name <small>i</small> :	Favorite <small>i</small> : No
Organization <small>i</small> :	Title <small>i</small> :
Email <small>i</small> :	Birthday <small>i</small> :
First Name <small>i</small> :	Family Name <small>i</small> :
Office Number:	Mobile Number:
Home Number:	
Sub Number<small>i</small>	
<input style="width: 100%; height: 150px; border: 1px solid #ccc; margin-bottom: 10px;" type="text"/>	
Sub Name:	Sub Number:
<input type="button" value="↑Add Sub"/>	
Note<small>i</small>	
<input style="width: 100%; height: 150px; border: 1px solid #ccc; margin-bottom: 10px;" type="text"/>	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

Figure 6-12

2) Upload Phonebook

You can upload a phonebook before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones. The format of phonebook should be *.xml.

Note:All the existing phonebooks of the IP phone will be replaced automatically if the phonebooks are configured in this way.

6.1.2.4 Configure phone

Let's take provisioning yealink as an example

Create New Phone have two modes,

Create New phone in webpage and Upload the IP Phone's configure file.

Add new phone via webpage

Click 'Add Phone' and fill in the corresponding information in the popup window

Add Phone

General	Codecs	Line Keys Settings
Enabled: Yes	NewConfig: No	
MAC Address: 001565	Name:	
Manufacturer: Yealink	Phone Type: T12	
Call Waiting: Enabled	Key As Send: #	
Auto Redial: Disabled	Auto Answer: Disabled	
Phone Book: Enabled		
Line <input type="checkbox"/> Line1 Extension: Label: Line Active: <input type="checkbox"/>		
<input checked="" type="button"/> Save <input type="button"/> Cancel		

Figure 6-13

1) General

· Enabled

Choose yes or no to enable or disable this extension

· NewConfig

If the firmware version number is greater than or equal to 70, please enable this option.

Note: when choose the Manufacturer as Yealink, a special option 'NewConfig' need to be considered. If the firmware version number of the Yealink phone is bigger or equal to 70, please enable this option.

· MAC address

Input the MAC address of IP phone

· Name

Put the name of this Phone here.

· Manufacturer

You can choose the Manufacturer of IP phone

· Phone Type

Choose the model of your phone. Only for snom phone

· Call Waiting

This call feature allows your phone to accept other incoming calls to an extension already in an active call.

·Key as Send

Configure the key as send, you choose # ,* or disable it

·Auto redial

Enable the auto redial for IP Phone

·Auto answer

Configure if auto answer is allowed for IP phone

·Phone book

Enable the feature of phone book of IP phone

2) Codecs

In this page, we can set the codecs for Ip phone

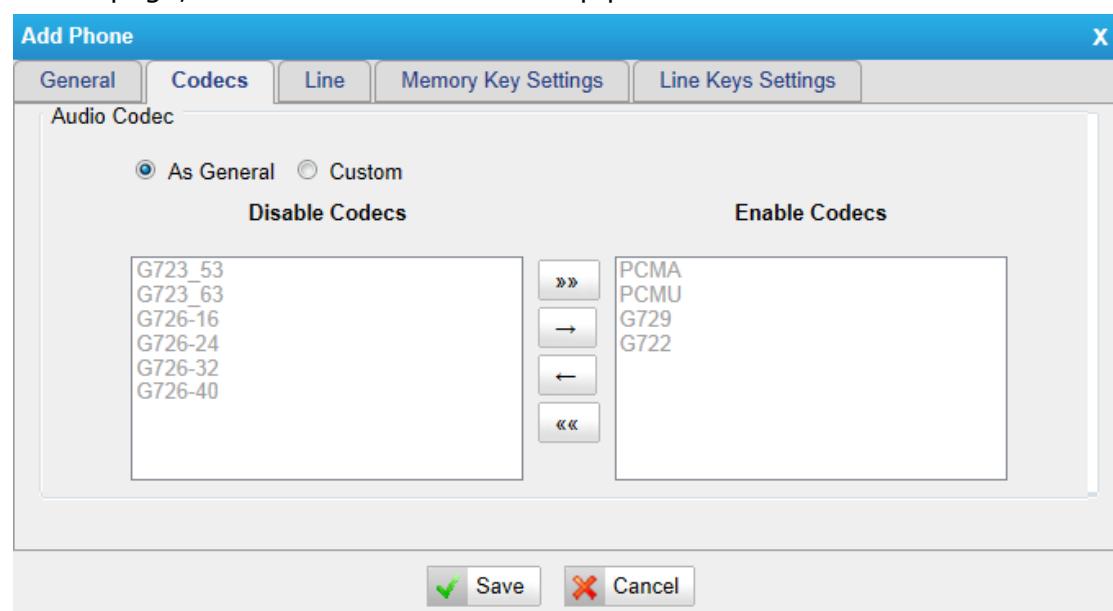


Figure 6-14

3) Line

In this page, we can set each line of IP phone for the account you want, active or not .

Add Phone

Line			
<input type="checkbox"/> Line1	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line2	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line3	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line4	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line5	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line6	Extension: <input type="text"/>	Label: <input type="text"/>	Line Active: <input type="checkbox"/>

 Save  Cancel

Figure 6-15

Extension: Selected the extension number for IP Phone.

Label: It is shown on the LCD for users to identify the account.

Line Active: You can choose on/off to enable/disable the account respectively

4) Memory key settings

In this page, we can configure the DSS keys of IP phone one by one .

Add Phone

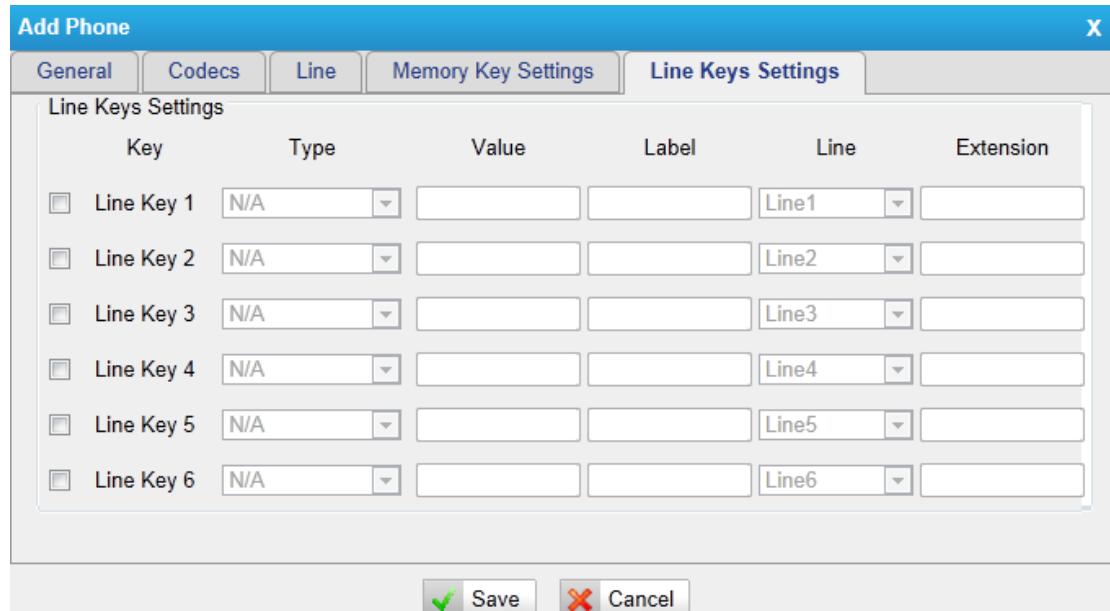
Memory Key Settings				
Memory Key				
Key	Type	Value	Line	Extension
<input type="checkbox"/> DSS Key1	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key2	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key3	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key4	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key5	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key6	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key7	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key8	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key9	N/A	<input type="text"/>	line1	<input type="text"/>
<input type="checkbox"/> DSS Key10	N/A	<input type="text"/>	line1	<input type="text"/>

 Save  Cancel

Figure 6-16

5) Line keys settings

We can configure the line key settings for this IP phone



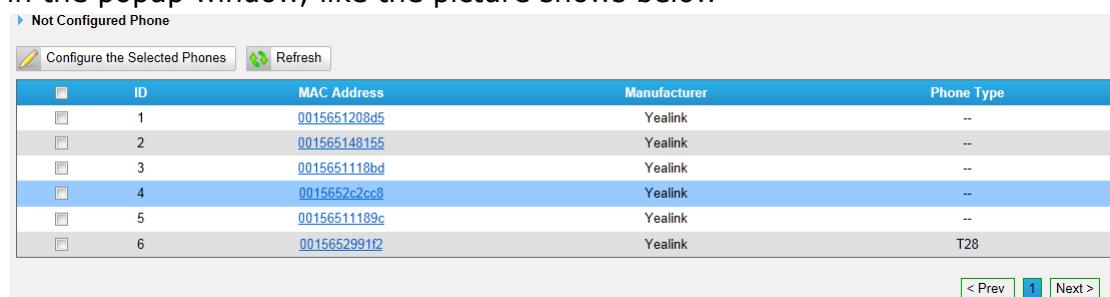
Key	Type	Value	Label	Line	Extension
Line Key 1	N/A			Line1	
Line Key 2	N/A			Line2	
Line Key 3	N/A			Line3	
Line Key 4	N/A			Line4	
Line Key 5	N/A			Line5	
Line Key 6	N/A			Line6	

✓ Save ✗ Cancel

Figure 6-17

6.1.2.5 Not configured phone

In this section, MyPBX will scan all the supported IP phones and display here, we can click the 'MAC address' of IP phone and input the corresponding information in the popup window, like the picture shows below



ID	MAC Address	Manufacturer	Phone Type
1	0015651208d5	Yealink	--
2	001565148155	Yealink	--
3	0015651118bd	Yealink	--
4	0015652c2cc8	Yealink	--
5	00156511189c	Yealink	--
6	0015652991f2	Yealink	T28

< Prev 1 Next >

Figure 6-18

6.1.2.6 Upload a file

Click 'Upload a file' and choose the configure file of IP phone in the popup window.

Note: the file format must be .cfg

Please edit the configuration files in advance before uploading



No Files Found.

Figure 6-19

6.2 Trunks

There are two types in MyPBX U300, the VOIP trunk and the E1 trunk, we cannot any other trunks like PSTN/GSM/BRI anymore.

6.2.1 Physical Trunk

In this page we can configure the details of E1 trunk, before configure anything, please make sure the cable is fine, and you have got enough information from the ISDN provider.



Physical Trunk			
E1/T1			
Trunk Name	Mode Type	Signaling	Echo Cancellation
E1Trunk1	E1	PRI-CPE	On

Figure 6-19

On the Physical Trunk page, click 'Edit' on the selected trunk and modify its properties in the popup window:

Edit Digital Trunk

Trunk Name:	E1Trunk1	Mode Type:	E1
Signaling:	PRI-CPE	PRI Dialplan:	unknown
Switch Type:	national	PRI Local Dialplan:	unknown
Reset Interval:	3600 s	PRI Indication:	Inband
Overlap Dial:	no	Nsf:	none
Enable Facility:	Enabled	Codec:	Default
CRC:	Off	Echo Training:	16ms
Echo Cancellation:	On	Caller ID Prefix	
		International Prefix:	
		Local Prefix:	
		Unknown Prefix:	
DOD Settings			
Global DOD:			
DOD:	601	Associated Extension:	601
<input type="button" value="Save"/> <input type="button" value="Cancel"/>			

Figure 6-20

• Trunk name

Set the trunk name

• Signaling

PRI-NET, PRI-CPE,MFC/R2,SS7

• Mode Type

E1 / T1 / J1

• Switch Type

national: National ISDN type2 (common in the US)

ni1: National ISDN type 1

dms100: Nortel DMS100

4ess: AT&T 4ESS

5ess: Lucent 5ESS

euroisdn: EuroISDN

qsig: Minimalistic protocol to build a "network" with two or more PBX of different

vendors!

·PRI Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most PRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Reset interval

Sets the time in seconds between restart of unused channels. Some PBXs don't like channel restarts, so set the interval to a very long interval e.g. 100000000 or 'never' to disable *entirely*. If you are in Israel, the following is important: As Bezeq in Israel doesn't like the B-Channel resets happening on the lines, it is best to set the reset interval to 'never' when installing a box in Israel. Our past experience also shows that this parameter may also cause issues on local switches in the UK and China.

·PRI Local Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Over Lap Dial

Define whether MyPBX can dial this switch using overlap digits. If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then MyPBX will wait after the last digit it receives.

·PRI Indication

Tells how Device should indicate Busy() and Congestion() to the switch/user. Accepted values are:

inband: Device plays indication tones without answering; not available on all PRI/BRI subscription lines .

outofband: Device disconnects with busy/congestion information code so the switch will play the indication tones to the caller. Busy() will now do same as setting PRI_CAUSE=17 and Hangup().

·Facility Enable

To enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility).

·NSF

Used with AT&T PRIs. If outbound calls are being rejected due to "Mandatory information element missing" and the missing IE is 0x20, then you need this

setting.

·CRC

CRC Verification.

·Codec

Default ,A-Law,U-Law.

·Echo Cancellation

Echo cancel Obviously this disables or enables echo cancellation, it is recommended to not turn this off.

1) CallerID Prefix

·International Prefix

When there are international calls coming in via this BRI trunk, the International Prefix you have set here will be added before the CID. So you can know this is an international call before you answer it.

·National Prefix

When there are national calls coming in via this BRI trunk, the National Prefix you have set here will be added before the CID. So you can know this is a national call before you answer it.

·Local Prefix

When there are Local calls coming in via this BRI trunk, the Local Prefix you have set here will be added before the CID. So you can know this is a local call before you answer it.

·Private Prefix

When there are Private calls coming in via this BRI trunk, the Private Prefix you have set here will be added before the CID. So you can know this is a Private call before you answer it.

·Unknown Prefix

When there are calls with unknown number coming via this BRI trunk, the Unknown Prefix you set here will be shown as the caller ID.

2) DOD Setting

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

·Global DOD

Global direct outward dialing number.

·DOD

Direct Outward Dialing Number.

·Associated Extension

The extension make call out via BRI Trunk will display the associated DOD.

6.2.2 VoIP Trunk

There are two types of VOIP trunk in MyPBX: SIP and IAX, in this page, we can also configure the 'service provider' trunk, which doesn't need the use name and password for authorization, when you have bought a trunk from provide with IP address only, please choose 'service provider' trunk .



Provider Name	Type	Hostname/IP	User Name
Yeastar	SIP	192.168.4.142	504

Provider Name	Type	Hostname/IP
Support.Tel	SIP	192.168.4.141

Figure 6-21

6.2.2.1 VoIP Trunk

In this page, we can configure VoIP trunk (SIP/ IAX) you have got from provider with the authorization name and password .

1) Add SIP Trunk

Input correct SIP information (provide by VoIP provider). Inaccurate information will prevent the trunk from registering.

Add VoIP trunk

Type: **SIP**

Provider Name:

Hostname/IP: : 5060

Domain:

User Name:

Authorization Name:

Password:

From User:

Online Number i:

Maximum Channels i: 0

Caller ID i:

Enable Outbound Proxy Server

Transport: **UDP** Enable SRTP i: Qualify:

DTMF Mode: **rfc2833**

DOD Settings

DOD: Associated Extension: **601**

Save Cancel

Figure 6-22

Type

SIP – Identifies whether the trunk sends and receives calls using the VoIP protocol SIP

Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar'.

Hostname/IP

Service provider's hostname or IP address. 5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.

Domain

VoIP provider's server domain name .

Username

Username of SIP account. Used for SIP trunk registration.

.Authorization name

Used for SIP authentication. Leave this blank if not required.

.Password

Password of SIP account .

.From User

All outgoing calls from this SIP Trunk will use the From User (In this case the account name for SIP Registration) in From Header of the SIP Invite package. Keep this field blank if not needed

.Online number

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if not needed.

.Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk . Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.

.Caller ID

Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'extension' screen will override the caller ID set in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

.Outbound Proxy Server

A proxy that receives requests from a client, even though it may not be the server resolved by the Request-URI.

.Codecs

Define the codec for this sip trunk and its priority

Note: To change the codec type and priority of this trunk, please create it first, it will appear when you edit it again.

.Transport

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

.Enable SRTP

Define if SRTP is enabled for this trunk

·Qualify

Send check alive packets to the sip provider.

·DTMF mode

Set default mode for sending DTMF of this trunk. Default setting: rfc2833

·DOD

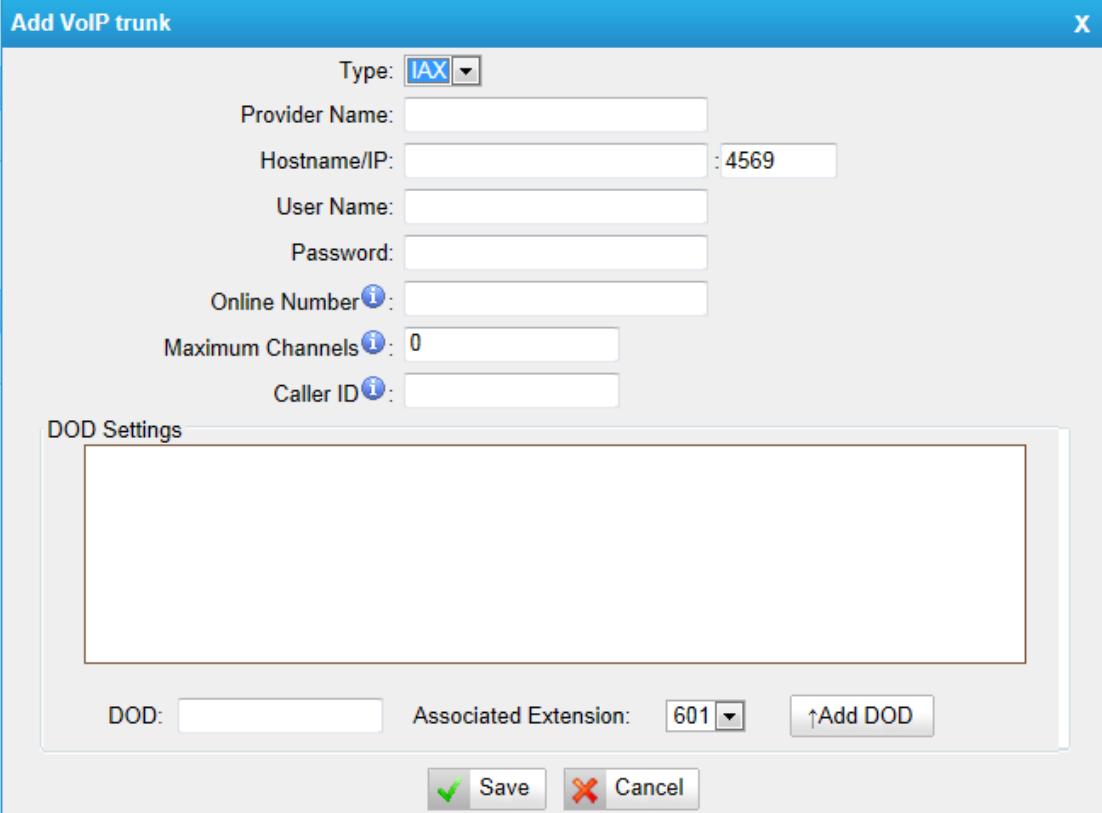
DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

·Associated Extension

The extension make call out via SIP Trunk will display the associated DOD

2) Add IAX trunk

Input correct IAX information (provided by VOIP provider). Inaccurate information will prevent the trunk from registering.



The screenshot shows the 'Add VoIP trunk' dialog box. At the top, it says 'Type: IAX'. Below that are fields for 'Provider Name', 'Hostname/IP:' (with ':4569' selected), 'User Name', 'Password', 'Online Number' (with a blue info icon), 'Maximum Channels' (set to 0), and 'Caller ID' (with a blue info icon). A large section titled 'DOD Settings' is present, which is currently empty. At the bottom, there are buttons for 'DOD:' (empty), 'Associated Extension:' (set to 601), and 'Add DOD'. There are also 'Save' and 'Cancel' buttons at the very bottom.

Figure 6-23

·Type

IAX – Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

·Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar2'.

·Hostname/IP

Service provider's hostname or IP address. 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

·Username

Username of IAX account. Used for IAX trunk registration.

·Password

Password of IAX account .

.Online number

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if it's no required.

·Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk . Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.

·Caller ID

Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'extension' screen will override the caller ID setting in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

·DOD

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

·Associated Extension

The extension make call out via IAX Trunk will display the associated DOD

6.2.2.2 Service Provider

This is service provider trunk (peer to peer mode), which authorized using IP address only. If you have got a trunk with IP address only, please choose this type

Edit Service Provider Trunk-SPS-Support.Tel

Type:	SIP
Provider Name:	Support.Tel
Hostname/IP:	192.168.4.141 : 5060
Maximum Channels <small>i</small> :	0
Codecs :	First: a-law Second: u-law Third: GSM Fourth: None Fifth: None
Transport:	UDP
Qualify:	<input checked="" type="checkbox"/>
DTMF Mode:	rfc2833
DOD Settings	
Global DOD:	
DOD:	Associated Extension: 601 <input type="button" value="↑Add DOD"/>

Figure 6-24

Type

SIP or IAX

SIP – Identifies whether the trunk sends and receives calls by using the VoIP protocol SIP.

IAX - Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

Provider Name

A unique label would help to you identify this trunk. Ex: 'Provider2'.

Hostname/IP

Service provider's hostname or IP address.

Note: 5060 is the standard port number used by SIP protocol, 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

Codecs

Define the codec for this sip trunk and its priority

Note: codec can only display when edit it after creating the trunk.

·Transport

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

·Qualify

Send check alive packets to the sip provider.

·DTMF mode

Set default mode for sending DTMF of this trunk. Default setting: rfc2833

·DOD

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

·Associated Extension

The extension make call out via this Trunk will display the associated DOD.

6.3 Outbound Call Control

6.3.1 Outbound Routes

In this page, we can configure the outbound rules to control the outgoing calls.

Note:

1. The max number of outbound route is 64
2. If the dial patterns are the same in several routes, MyPBX will choose the available routes from top to the last one.
3. When you have created a new extension, please edit the outbound route so that he can dial out too

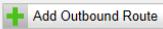
Outbound Routes	
 Add Outbound Route	
Route Name	Dial Pattern
pstnout	9.

Figure 6-25

We can create outbound route or use the default route 'pstnout' (dial 9+numbers to dial out)

Edit Outbound Route - pstnout

Route Name <small>i</small> :	<input type="text" value="pstnout"/>
Dial Pattern <small>i</small> :	<input type="text" value="9 "/>
Strip <small>i</small> :	1 digits from front
Prepend these digits <small>i</small> :	<input type="text"/> before dialing
Password:	<input type="text"/>
T.38 Support <small>i</small> :	No
Rrmemory Hunt <small>i</small> :	No

Member Extensions i

Available Extensions	Selected
<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 601(FXS) 602(FXS)

Member Trunks i

Available Trunks	Selected
<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	E1Trunk1(E1)

Figure 6-26

•Route Name

Name of this Outbound Route. Ex: 'Local' or 'Long Distance' etc.

•Dial Pattern

Outbound calls that match this dial pattern will use this outbound route. There are a number of dial pattern characters that have special meanings:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9) The '.' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Strip digits from front

Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.

·Prepend these digits before dialing

These digits will be prepended to the phone number before the call is placed. For example, if a trunk requires 10 digit dialing, but users are more comfortable with 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit phone numbers before calls are placed.

·Password

The route password can be used to protect this route from being accessed without a password.

·T.38 Support:

Enable T38 fax in this outbound route(Only for SIP Trunk).

·Rrmemory Hunt

Round robin with memory, remembers which trunk was used last time, and then use the next available trunk to call out.

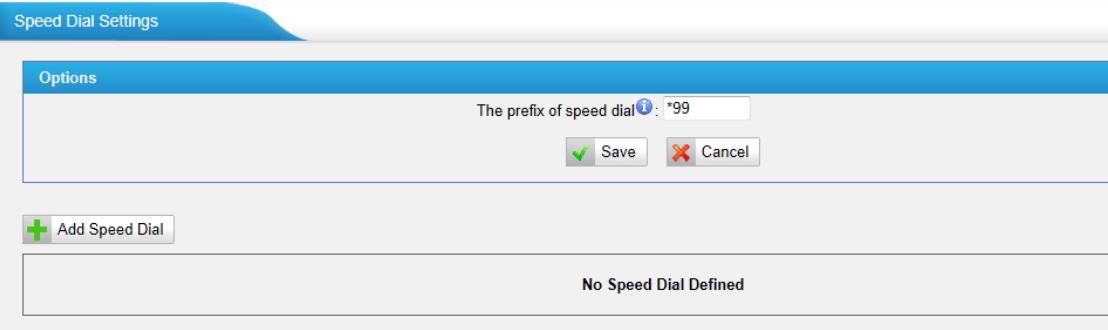
·Member Extensions

Defines the extensions that will be permitted to use this outbound route.

·Member Trunks

Defines the trunks that can be used for this outbound route.

6.3.2 Speed Dial Settings



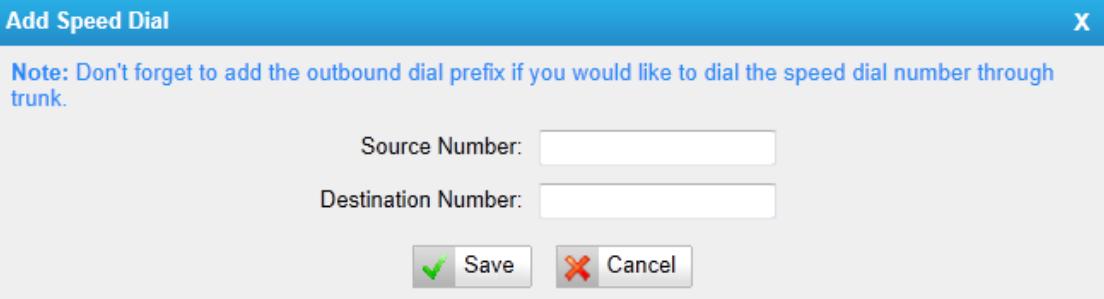
The screenshot shows a web-based configuration interface for 'Speed Dial Settings'. At the top, there's a blue header bar with the title 'Speed Dial Settings'. Below it is a section titled 'Options' with a sub-instruction 'The prefix of speed dial (I): *99'. There are two buttons at the bottom of this section: a green 'Save' button with a checkmark icon and a red 'Cancel' button with a cross icon. Below the 'Options' section is a button labeled '+ Add Speed Dial' with a green plus sign icon. At the very bottom of the page, a message reads 'No Speed Dial Defined'.

Figure 6-26

1) Options

The prefix of speed dial

The prefix should be dialed before the speed dial number. Default is *99



Add Speed Dial

Note: Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

Source Number:

Destination Number:

Save Cancel

Figure 6-27

2) Add new speed dial.

Source Number

The speed dial number.

Destination Number

The number you want to call.

e.g. The source number is "123". The destination number is 5503305. The prefix number is *99. You can use an extension with any type to dial *99123, then it will call to number 5503305.

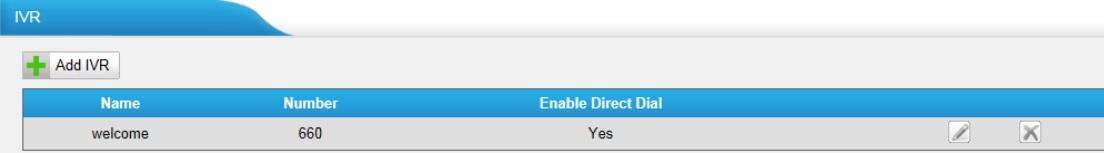
Note: Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

6.4 Inbound Call Control

In this page, we can configure the details of IVR, ring group, queue and inbound routes.

6.4.1 IVR

When there's an inbound call aims at Auto Attendant, MyPBX will play an IVR recording and route the caller to the requested destination (for example, 'Welcome to XX company, for sales press 1, for technical support press 2, for operator press 0, etc'). The system will transfer the call to corresponding extension according to DTMF digits inputted by the user



Name	Number	Enable Direct Dial
welcome	660	Yes

Figure 6-28

There is a default IVR here, we can edit it directly or add IVR by yourself

Edit IVR - welcome

Number <small>i</small> :	660																																													
Name <small>i</small> :	welcome																																													
Prompt <small>i</small> :	default <input type="button" value="Custom Prompts"/>																																													
Repeat Count <small>i</small> :	3 <input type="button"/>																																													
Key Timeout <small>i</small> :	3 <input type="button"/>																																													
<input checked="" type="checkbox"/> <small>i</small> Enable Direct Dial																																														
Keypress Events <table border="1"> <thead> <tr> <th>Key</th> <th>Action</th> <th>Destination</th> </tr> </thead> <tbody> <tr> <td>0</td> <td>Connect to Extension <input type="button"/></td> <td>Extension -- 300 <input type="button"/></td> </tr> <tr> <td>1</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>2</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>3</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>4</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>5</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>6</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>7</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>8</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>9</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>#</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>*</td> <td>No Action <input type="button"/></td> <td><input type="button"/></td> </tr> <tr> <td>Timeout <small>i</small></td> <td>Connect to Extension <input type="button"/></td> <td>Extension -- 300 <input type="button"/></td> </tr> <tr> <td>Invalid <small>i</small></td> <td>Connect to Extension <input type="button"/></td> <td>Extension -- 300 <input type="button"/></td> </tr> </tbody> </table>		Key	Action	Destination	0	Connect to Extension <input type="button"/>	Extension -- 300 <input type="button"/>	1	No Action <input type="button"/>	<input type="button"/>	2	No Action <input type="button"/>	<input type="button"/>	3	No Action <input type="button"/>	<input type="button"/>	4	No Action <input type="button"/>	<input type="button"/>	5	No Action <input type="button"/>	<input type="button"/>	6	No Action <input type="button"/>	<input type="button"/>	7	No Action <input type="button"/>	<input type="button"/>	8	No Action <input type="button"/>	<input type="button"/>	9	No Action <input type="button"/>	<input type="button"/>	#	No Action <input type="button"/>	<input type="button"/>	*	No Action <input type="button"/>	<input type="button"/>	Timeout <small>i</small>	Connect to Extension <input type="button"/>	Extension -- 300 <input type="button"/>	Invalid <small>i</small>	Connect to Extension <input type="button"/>	Extension -- 300 <input type="button"/>
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Timeout <small>i</small>	Connect to Extension <input type="button"/>	Extension -- 300 <input type="button"/>																																												
Invalid <small>i</small>	Connect to Extension <input type="button"/>	Extension -- 300 <input type="button"/>																																												
<input checked="" type="checkbox"/> <small>i</small> Save <input type="button" value="Cancel"/>																																														

Figure 6-29

·Number

MyPBX treats IVR as an extension; you can dial this extension number to reach the IVR from internal extension.

·Name

A name for the IVR

·Prompt

The prompt recording that will be played when this IVR is reached.

·Repeat Count

The number of times that the selected IVR prompt will be played.

·Key Timeout

Wait for the user to enter a new extension for a specified number of seconds.

·Enable Direct Dial

Allow the caller to dial other extensions number directly.

·Key Press Events

A list of actions that can be performed depending on the digit dialed by the user.

·Key

The Key pressed when the callers hear the IVR prompt.

·Action

When the callers press the corresponding key, the action MyPBX executes.

No Action: Do nothing

Connect to Extension: Connect the call to an extension.

Connect to Voicemail: Connect the call to the voicemail of an extension

Connect to RingGroup: Connect the call to a ringgroup.

Connect to IVR: Connect the call to an IVR.

Connnect to Conference Room: Connect the call to a conference room.

Connect to DISA: Connect the call to a DISA.

Connect to Queue: Connect the call to a queue.

Connect to Faxes: Connect the call to Faxes of extensions.

Dial by Name: The callers can dial the name of an extension to connect to the corresponding extension.

Hung up: Hung up the call.

·Destination

Where will MyPBX route the call when the action occurs.

·Time Out

Defines the timeout action. A timeout occurs after the IVR prompt has finished playing for the number of times specified by the 'Repeat Count' field.

·Invalid

Defines the invalid action. The invalid action is triggered if the user enters a DTMF digit that is not defined for this IVR.

6.4.2 Ring Groups

Ring groups can be configured to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

Note: follow me feature in extension page will not take effect when it's ringing as an agent

Ring Groups		
 Add Ring Group		
Number	Name	Members
620	ringgroup_default	300(SIP)-301(SIP)-302(SIP)-303(SIP)-304(SIP)-305(S...)

Figure 6-30

There is a default ringgroup, you can edit it or create a new one

Edit Ring Group - ringgroup_default

Ring Group Name <small>i</small> :	<input type="text" value="ringgroup_default"/>												
Ring Group Number <small>i</small> :	<input type="text" value="620"/>												
Strategy <small>i</small> :	<input type="button" value="Ring all simultaneously"/>												
Seconds to ring each member <small>i</small> :	<input type="text" value="60"/>												
Ring Group members <small>i</small> <table style="width: 100%; border-collapse: collapse;"> <tr> <th style="width: 30%;">Available Extensions</th> <th style="width: 10%; text-align: center;">»»</th> <th style="width: 60%;">Selected</th> </tr> <tr> <td style="border: 1px solid #ccc; padding: 5px; height: 100px; vertical-align: top;"> 601(FXS) 602(FXS) </td> <td style="border: 1px solid #ccc; width: 10px; text-align: center; background-color: #f0f0f0; height: 100px; vertical-align: top;">     </td> <td style="border: 1px solid #ccc; padding: 5px; height: 100px; vertical-align: top;"> 300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) </td> </tr> </table>		Available Extensions	»»	Selected	601(FXS) 602(FXS)	   	300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP)						
Available Extensions	»»	Selected											
601(FXS) 602(FXS)	   	300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP)											
Destination If No Answer: <table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%;">Destination:</td> <td style="width: 70%;"> <input checked="" type="radio"/> End Call <input checked="" type="radio"/> Extension <input type="button" value="Extension -- 300"/></td> </tr> <tr> <td></td> <td><input type="radio"/> Voicemail <input type="button" value="Voicemail -- 601"/></td> </tr> <tr> <td></td> <td><input type="radio"/> IVR <input type="button" value="IVR -- welcome"/></td> </tr> <tr> <td></td> <td><input type="radio"/> Ring Group <input type="button" value="Ring Group -- ringgroup"/></td> </tr> <tr> <td></td> <td><input type="radio"/> Conference Room <input type="button" value="Conference Room -- 64"/></td> </tr> <tr> <td></td> <td><input type="radio"/> Queues <input type="button" value="Queues --"/></td> </tr> </table>		Destination:	<input checked="" type="radio"/> End Call <input checked="" type="radio"/> Extension <input type="button" value="Extension -- 300"/>		<input type="radio"/> Voicemail <input type="button" value="Voicemail -- 601"/>		<input type="radio"/> IVR <input type="button" value="IVR -- welcome"/>		<input type="radio"/> Ring Group <input type="button" value="Ring Group -- ringgroup"/>		<input type="radio"/> Conference Room <input type="button" value="Conference Room -- 64"/>		<input type="radio"/> Queues <input type="button" value="Queues --"/>
Destination:	<input checked="" type="radio"/> End Call <input checked="" type="radio"/> Extension <input type="button" value="Extension -- 300"/>												
	<input type="radio"/> Voicemail <input type="button" value="Voicemail -- 601"/>												
	<input type="radio"/> IVR <input type="button" value="IVR -- welcome"/>												
	<input type="radio"/> Ring Group <input type="button" value="Ring Group -- ringgroup"/>												
	<input type="radio"/> Conference Room <input type="button" value="Conference Room -- 64"/>												
	<input type="radio"/> Queues <input type="button" value="Queues --"/>												
<input type="button" value="Save"/> <input type="button" value="Cancel"/>													

Figure 6-31

·Ring Group Name

This option defines a name for this group, i.e. 'Sales'. 'Ring Group Name' is a label to help you identify this group in the group list.

·Ring Group Number

This option defines the numbered extension that can be dialed to reach this group.

·Strategy

This option sets the Ringing Strategy for this Group. The options are as follows:

1. RingAll Simultaneously: Ring all available Extensions simultaneously.
2. Ring Sequentially: Ring each extension in the group one at a time.

·Seconds to ring each member

1. If the strategy is 'Ring All Simultaneously', it means set the number of seconds to ring this group before routing the call according to the 'Destination if No Answer' settings.
2. If the strategy is 'Ring Sequentially', it means set the number of seconds to ring a single extension before moving onto the next one.

· Ring Group Members

An extension can be made a member of this ring group by moving it into the 'Selected' box.

·Destination If No Answer

When all members on this group fail to answer the call, system will handle the call according to the selected destination.

6.4.3 Queues

Call Queues give users (i.e. call centers) an efficient means to have their calls answered in the order they were received to deliver top tier customer service.

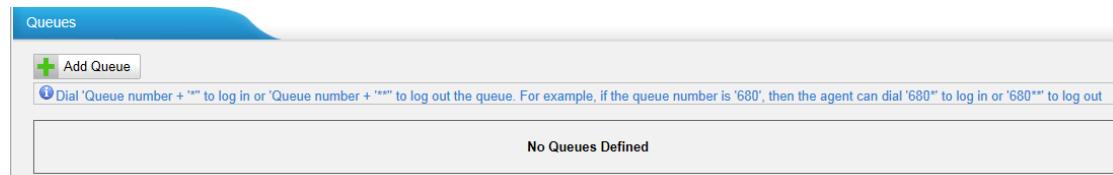


Figure 6-32

Call queues allow calls to be sequenced to one or more agents.

Note:

1. Dial 'Queue number + *' to log in or 'Queue number + **' to log out the queue. For example, if the queue number is '680', then agent can dial '680*' to log in or '680**' to log out.
2. Follow me feature in extension page will not take effect when it's ringing as an agent of queue

Add Queue

Queue Name <small>i</small> :	680				
Queue Number <small>i</small> :	680				
Queue Password <small>i</small> :					
Queue Agent Timeout <small>i</small> :	30				
Queue Max Wait Time <small>i</small> :	1800				
Queue Ring Strategy <small>i</small> :	ringall				
Agents <small>i</small>					
<table border="1"> <thead> <tr> <th>Available Agents</th> <th>Selected</th> </tr> </thead> <tbody> <tr> <td>300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 601(FXS) 602(FXS)</td> <td></td> </tr> </tbody> </table>		Available Agents	Selected	300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 601(FXS) 602(FXS)	
Available Agents	Selected				
300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 601(FXS) 602(FXS)					
Caller Position Announcements					
Announce Position <small>i</small> : Yes <input type="checkbox"/> Announce Hold Time <small>i</small> : Yes <input type="checkbox"/> Frequency <small>i</small> : 30 seconds <input type="button"/>					
Periodic Announcements					
Prompt <small>i</small> : <input type="text"/> Custom Prompts Frequency <small>i</small> : 30 seconds <input type="button"/>					
Events					
Key: --- <input type="button"/> Action: End Call <input type="button"/> Destination: <input type="button"/>					
Failover-Destination					
Action: End Call <input type="button"/> Destination: <input type="button"/>					
Others					
Music On Hold <small>i</small> : calmriver <input type="button"/> Music on Hold Prompts Leave When Empty <small>i</small> : Yes <input type="checkbox"/> Join Empty <small>i</small> : No <input type="checkbox"/> Agent Announcement <small>i</small> : <input type="text"/> Join Announcement <small>i</small> : <input type="text"/> Retry <small>i</small> : 30 Wrap-up Time <small>i</small> : 30					
<input checked="" type="button"/> Save <input type="button"/> Cancel					

Figure 6-33

·Queue Name

A name for the Queue.

·Queue Number

Use this number to dial into the queue, or transfer callers to this number to put them into the queue.

·Queue Password

You can require agents to enter a password before they can login to this queue. This field allowed to be blank empty, it means the agents can login to this queue without password.

·Queue Agent Timeout

The number of seconds an agent's phone can ring before we consider it a timeout.

·Queue Max Wait Time

The maximum number of seconds a caller can wait in a queue before being pulled out. (0 for unlimited).

·Queue Ring Strategy

This option sets the Ringing Strategy for this Queue. The options are

RingAll: Ring All available Agents simultaneously until one answers.

LeastRecent: Ring the Agent which was least recently called.

FewestCalls: Ring the Agent with the fewest completed calls.

Random: Ring a Random Agent.

RRmemory: Round Robin with Memory, Remembers where it left off in the last ring pass".

1) Agents

This selection shows all users. Selecting a user here makes them a agent of the current queue.

2) Caller Position Announcements

·Announce Position

Announce position of caller in the queue

·Announce Hold Time

Enabling this option causes MyPBX to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

·Frequency

How often to announce queue position and estimated hold time.

Note: '0 seconds' means disable the announcement

3) Periodic Announcements

·Prompt

Select a prompt file to play periodically.

·Frequency

How often to announce a prompt to the caller.

4) Events

If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.

5) Failover-Destination

Define the failover action. A failover occurs after the user reaches the Queue max wait time.

6) Others

·Music On Hold

Select the 'Music on Hold' Class for this Queue.

·Leave When Empty

This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.

Yes: Callers are forced out of a queue when no agents are logged in.

No: Callers will remain in a queue with no agents.

·Join Empty

This option controls whether callers can join a call queue that has no agents.

There are two options,

Yes: Callers can join a call queue with no agents or only unavailable agents

No: Callers cannot join a queue with no agents

The default option is No.

·Agent Announcement

Announcement played to the Agent prior to bridging in the caller.

·Join Announcement

Announcement played to callers once prior to joining the queue.

·Retry

The number of seconds we wait before trying all the phones again.

·Wrap-up time

How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 30.

6.4.4 Conferences

Conferences			
Conference Room	Admin	PIN #	
640	--	No Password	 
641	--	No Password	 
642	--	No Password	 
643	--	No Password	 
644	--	No Password	 

Figure 6-34

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. Conference agents can dial * to access to the settings options and the admin can kick the last user out and can lock the conference room.

·Extension

This is the number dialed to reach this Conference Room.

·Admin

Admin can kick a user out and can lock the conference room.

·Pin

Set a PIN # that must be entered in order to access this conference room (i.e. 1234).

Edit Conference Room 640

Extension i:

Admin i:

PIN # i:

Save
Cancel

Figure 6-35

6.4.5 Inbound Routes

Inbound routing processes incoming call traffic to destination extensions during office hours or outside office hours

Inbound Routes		
 Add Inbound Route	Route Name	DID Number
		Caller ID Number
	pstn9n	

Figure 6-36

There is a default inbound route for all the trunks and set IVR as the destination, you can edit it or create a new one for your demands. When an incoming call arrives, the system will first check 'fax detection', then 'Holidays', at last 'Business Days'.

Edit Inbound Route: pstn9n

General	
Route Name  :	<input type="text" value="pstn9n"/>
DID Number  :	<input type="text"/>
Extension  :	<input type="text"/>
Caller ID Number  :	<input type="text"/>
Distinctive Ringtone  :	<input type="text"/>
Enable Callback :	No <input type="button" value="Callback Settings"/>
Member Trunks  	
Available Trunks	
Yeastar(SIP) 1(IAX) Support.Tel(SPS)	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>
Selected	
E1Trunk1(E1)	
Business Days	
Office Hours :	<input type="button" value="default"/>
Office Hours Destination :	<input type="button" value="IVR"/> <input type="button" value="IVR -- welcome"/>
Non-office Hours Destination :	<input type="button" value="IVR"/> <input type="button" value="IVR -- welcome"/>
During Holidays	
Holiday :	<input type="button"/>
Destination :	<input type="button" value="End Call"/> <input type="button"/>
Fax Detection	
Destination :	<input type="button" value="No Detect"/> <input type="button"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

figure 6-37

1) General

·Route Name

A name for this inbound route. Ex: 'pstn9n' etc.

·DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. You can also use pattern matching to match a range of numbers. The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX**will match any 7 digits phone number.

Example 2: **1NXXNXXXX**will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

For more information, please refer to [**Appendix G How to Use DID.**](#)

·Extension

Define the extension for DID number. This field is only valid when you use BRI, SIP, SPS or SPX trunk for this inbound router. You can only input number and '-'in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

·Caller ID Number

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no DID info.

You can also use a pattern match (e.g. 2[345]X) to match a range of numbers.

The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX**will match any 7 digits phone number.

Example 2: **1NXXNXXXX**will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Distinctive Ringtone

MyPBX support mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to '**Family**', the ring tone will be played if the phone receives the incoming call. This feature works with all IP phones which support distinctive ringtone.

How do I configure distinctive ring tones? Please refer to [APPENDIX E](#).

Note: If you want to use feature Distinctive ringtone, please confirm your IP phone support this feature firstly. Currently distinctive ringtone can be compatible with Yealink and Snom phone.

·Enable Callback

You can enable the callback function of this inbound route. If you want to configure the callback function, please refer to [chapter 3.5.12](#)

2) Member Trunks

This area allows you to select which trunks will be member trunks for this route. To make a trunk a member of this route, please move it to the 'Selected' box.

3) Business Days

Define where the calls will be routed during Business Days.

·Office Hours

Select one defined business days office hours.

·Office Hours Destination

Configure where to route the incoming calls during office hours.

·End Calls

Route the incoming calls to end calls, System will auto hang-up the call.

·Extension

Route the incoming calls to a specific extension.

·Voicemail

Route the incoming calls to extension's voicemail.

·IVR

Route the incoming calls to a specific IVR.

·Ring Group

Route the incoming calls to a specific Ring Group.

·Conference Room

Route the incoming calls to a specific Conference Room.

·DISA

Route the incoming calls to a specific DISA.

·Queues

Route the incoming calls to a specific Queue.

·Faxes

Route the incoming faxes to a specific extension's mail address.

Note: This function only supports T.38 faxes.

·Outbound Routes

Route the incoming calls to a specific outbound route.

This function is mainly used for the connection of two branches.

For example: Company A locates headquarters in the USA with a branch B in China. A and B both have MyPBX phone systems.

Now if staff of A would like to make a call to a telephone or mobile phone in China from the extension of A but via the FXS line of B, that can be done by this configuration.

·Non-office Hours Destination

Configure where to route the incoming calls during non-office hours.

4) During Holidays

Define where the calls will be routed during Holidays.

·Holiday

Select the defined Holiday to use. When a time is defined in both Business Days and Holidays, it will be treated as Holidays.

·Destination

Configure where to route the incoming calls during holidays.

5) Fax Detection

Configure if detecting faxes in this inbound route.

Note: Please choose IVR as the destination above before configure fax detection (recommend)

·Destination

Configure where the faxes will be routed when faxes are detected.

·No detect

Do not detect faxes.

•Custom Email

Customize an E-mail address to receive the faxes. You should first configure the 'Voicemail Settings->SMTP Settings for Voicemail' correctly before you use this option.

•Faxes

Send faxes to an extension. If choosing a FXS extension here, the fax will be sent to the FXS port selected, you should connect a fax machine to this FXS port. If Choosing a VoIP extension, the fax will be sent to the extension's voicemail as an attachment.

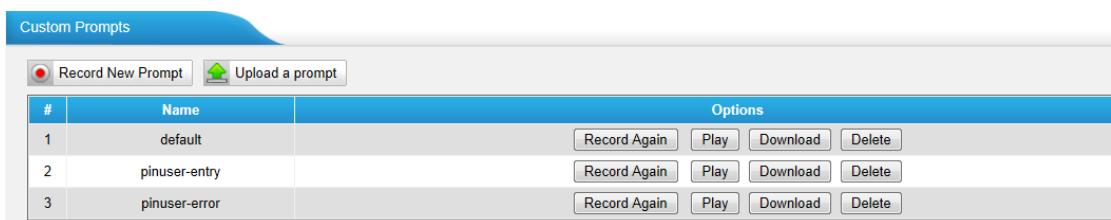
Note: If you receive faxes with custom Email address, the 'SMTP settings' of 'Voicemail Settings' should be configured successfully in advance. If you receive faxes with E-mail address configured in VOIP extension voicemail, you should first make sure the tested email to your email address works fine.

6.5 Audio Settings

It's allowed to customize the prompts in MyPBX, including the Audio In and change the system prompts to your local country.

6.5.1 Custom Prompts

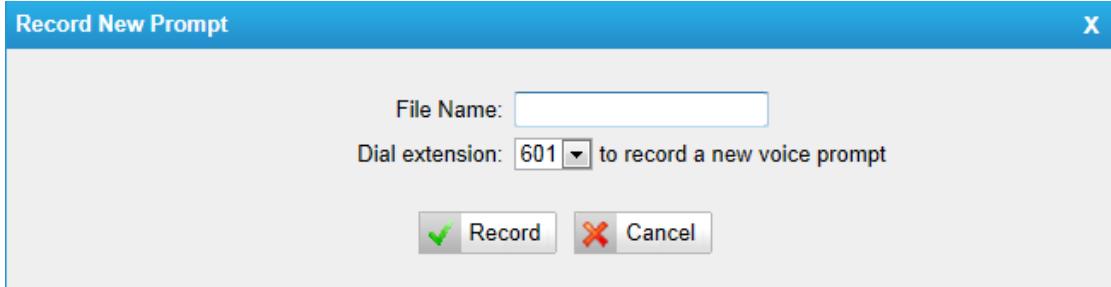
We can record or upload the prompts in this page, you can also play it directly to confirm if it's a valid one, you can also download it and save it as an backup



Custom Prompts		
#	Name	Options
1	default	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
2	pinuser-entry	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
3	pinuser-error	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>

Figure 6-38

1. Record new Prompt



Record New Prompt

File Name:

Dial extension: to record a new voice prompt

Figure 6-39

The administrator can use this screen to record custom prompts by doing the following:

- 1) Click 'Record New Custom Prompt'
- 2) Input the desired file name on the popup window and choose an extension to call for recording (such as 500).
- 3) Click 'Record'. The selected extension will ring and you can pick up the phone to start recording.

2. Upload Prompt



Figure 6-40

The administrator can also upload prompts by doing the following:

- 1) Click 'Upload Prompt'.
- 2) Click 'Browse' to choose the desired prompt.
- 3) Click 'Upload' to upload the selected prompt.

Note: The file size must not be larger than 1.8MB, and the file must be WAV format:

GSM 6.10 8kHz, Mono, 1Kb/s;
Alaw/Ulaw 8kHz, Mono, 1Kb/s;
PCM 8kHz, Mono, 16Kb/s.

6.5.2 Music on Hold Prompts

In this page, we can upload the music on hold prompts or adjust the volume from Audio In interface (Only available in MyPBX U300)



Figure 6-41

The administrator can upload on hold music as follows:

- 1) Click 'Upload Music on Prompt'
- 2) Click 'Browse' to choose the desired audio file.
- 3) Click 'Upload' to upload the selected file.



Figure 6-42

Note: The sound file format should be as follows:

GSM 6.10, 8.000kHz, Mono, 1kb/sec;

Alaw/Ulaw 8.000kHz, Mono, 1kb/sec;

PCM 8.000kHz, Mono, 16kb/sec.

The file size must not be greater than 1.8MB.

6.5.3 System Prompts Settings

MyPBX have prompts of many languages. You can download the appropriate language you need. MyPBX can support American English, Australian English, Chinese, Dutch, French, Canadian French, German, Greek, Hungarian, Italian, Polish, Portuguese, Brazilian Portuguese, Russian, Spanish, Mexican Spanish, Turkish, Thai, Korean, Danish, Finnish, Norwegian, Swedish, Arabic, British currently.

Note:

1. Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server, please contact the local dealer for the prompts
2. When update successfully, just click 'apply the changes' on web then it will take effect, there is no need to reboot.

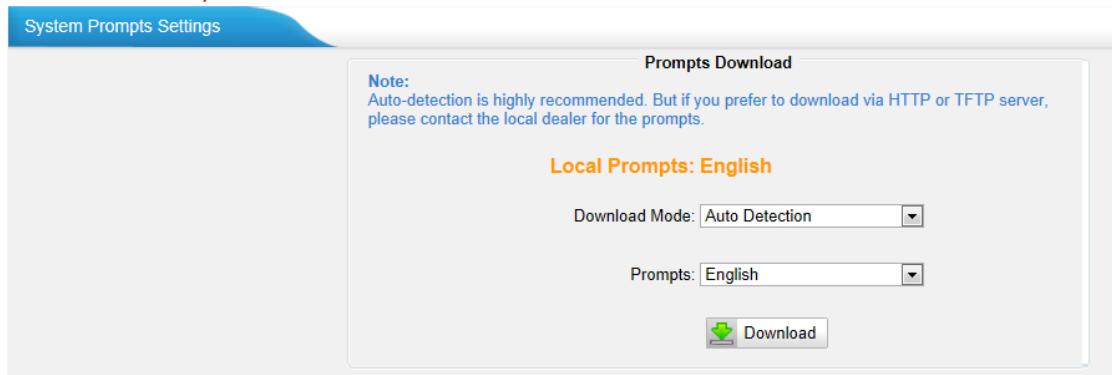


Figure 6-43

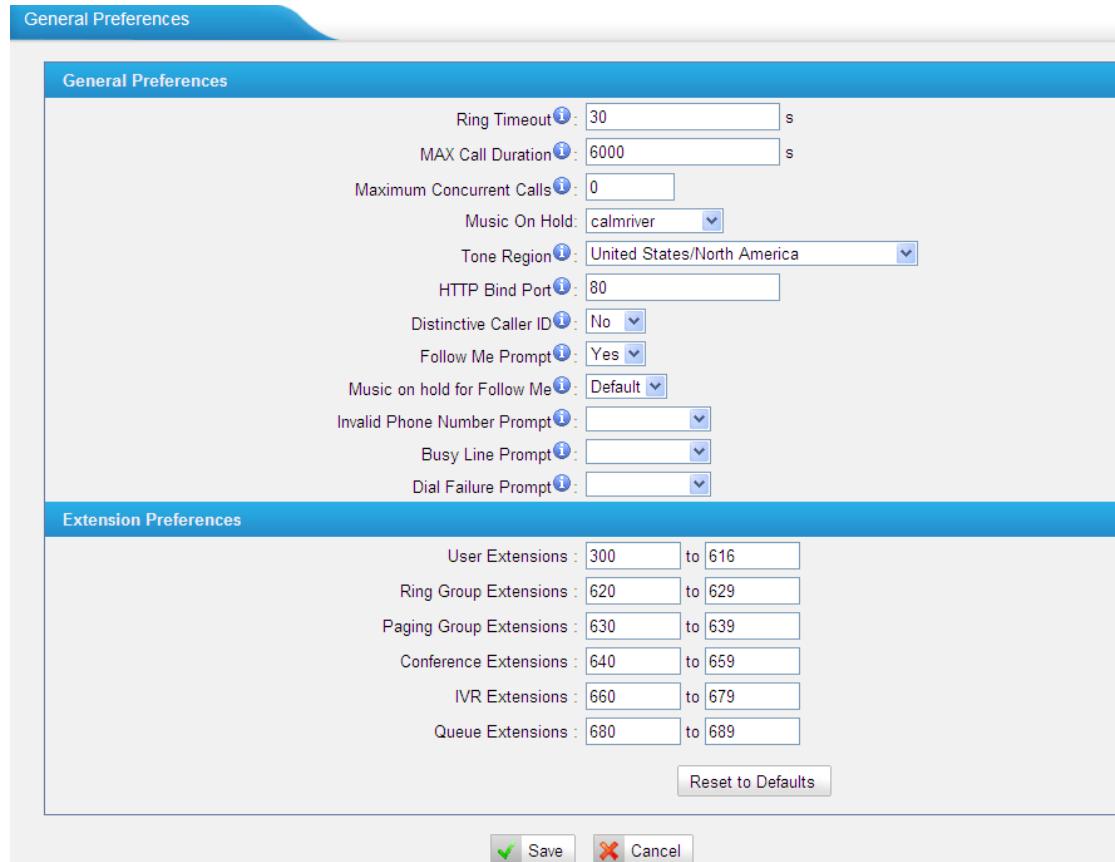
6.6 Basic Settings

There are some basic settings we need to configure MyPBX U300, like the

general preferences, business hours, feature codes, voicemail settings

6.6.1 General Preferences

In this page, there are some general settings of MyPBX



General Preferences

- Ring Timeout: 30 s
- MAX Call Duration: 6000 s
- Maximum Concurrent Calls: 0
- Music On Hold: calmriver
- Tone Region: United States/North America
- HTTP Bind Port: 80
- Distinctive Caller ID: No
- Follow Me Prompt: Yes
- Music on hold for Follow Me: Default
- Invalid Phone Number Prompt:
- Busy Line Prompt:
- Dial Failure Prompt:

Extension Preferences

User Extensions:	300	to	616
Ring Group Extensions:	620	to	629
Paging Group Extensions:	630	to	639
Conference Extensions:	640	to	659
IVR Extensions:	660	to	679
Queue Extensions:	680	to	689

Figure 6-44

1) General

·Ring Timeout

Number of seconds to ring a device before handling the call as per the extension's Follow Me settings . Default value is 30s.

·MAX call duration

The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. Default value is 6000s.

·Maximum concurrent calls

Maximum concurrent calls limits. Default value 0 means no limit

·Music on hold

Used to set hold music for the system.

Note: if you need use the 'live music' from 'Audio In' interface, please choose ' external music' here.

·Tone Region

Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.

Note: please reboot the system to take it effect.

·HTTP bind port/Web Access Port

Port to use for HTTP sessions. Default: 80

Note: please reboot the system to take it effect.

· Distinctive Caller ID

When incoming calls are routed from ring group/queue/IVR, the caller ID displays with the name of ring group/queue/IVR , for example 5503302(ringgroup_default)

Note: To display IVR's name, please press the key instead of the extension number directly.

·Follow Me Prompt

Configure whether to play a prompt 'please wait while trying to look at the person you are calling' when transfer a call by follow me settings.

·Music on hold for Follow Me

Choose to play default music-on-hold, ringing tone or stay mute while 'Follow Me'.

·Invalid Phone Number Prompt

Configure the prompt when the dialed phone number is invalid.

·Busy Line Prompt

Configure the prompt when the dialed phone number is busy.

·Dial Failure Prompt

Configure the prompt when dial failed due to conjunction no-available channel.

2) Extension Preferences

·User Extensions

The default value is 500 to 616

·Ring Group Extensions

The default value is 620 to 629

·Paging Group Extensions

The default value is 630 to 639

·Conference Extensions

The default value is 640 to 659

·IVR Extensions

The default value is 660 to 679

·Queue Extensions

The default value is 680 to 689

6.6.2 Business Hours

Business hours including 'holiday configuration' is used to control the incoming calls, we can configure it in this page.

Figure 6-45

1) General

- Enable Business Hours**
- disable Business Hours**

2) Others

·Enable Office Closed Timing

By dialing *81 (*81 is default) on an extension will force the office time closed for the device whatever the general setting is.

·Enable Office Timing

By dialing *82 (*82 is default) on an extension will force the office time enabled for the device whatever the general setting is.

·Disable Office closed timing

By dialing *081 (*081 is default) on an extension will disable the Office Closed

Timing.

3) Add office hours

You can setup the business hours here.

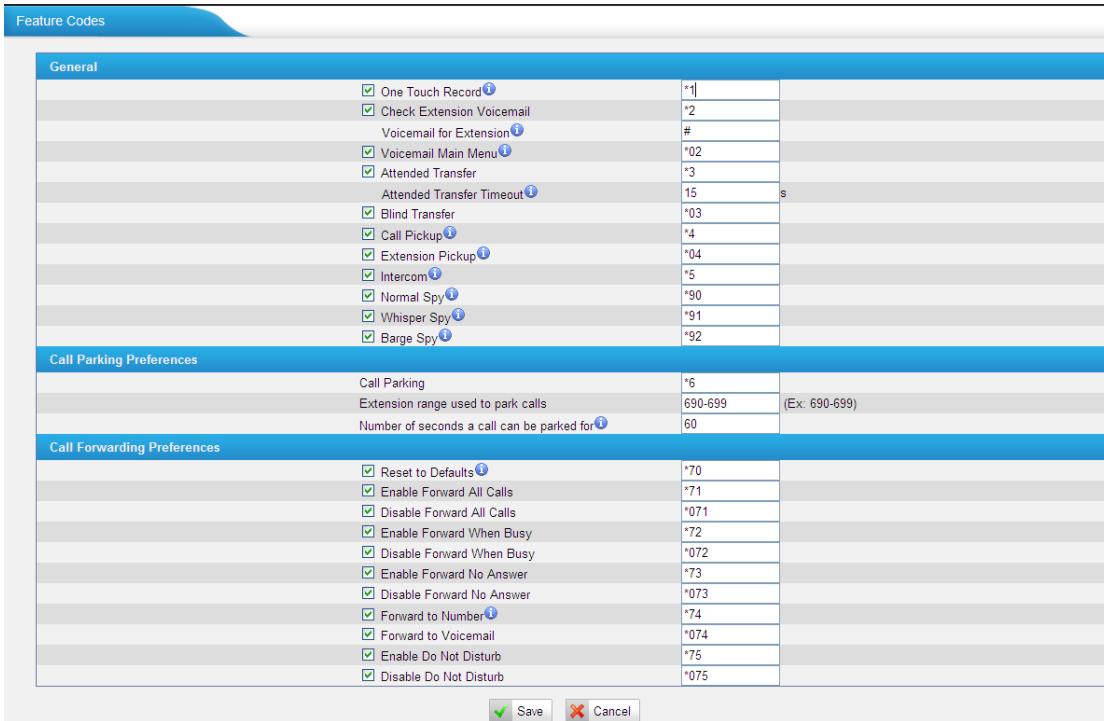
4) Add Holiday

You can setup the holidays here.

If a time period is configured as both Holidays and office hours, it will be treated as Holidays.

6.6.3 Feature Codes

There are many feature codes available in MyPBX, which allow users to dial from extension side to realize the exact feature



General	
<input checked="" type="checkbox"/> One Touch Record <small>①</small>	*1
<input checked="" type="checkbox"/> Check Extension Voicemail	*2
VoiceMail for Extension <small>①</small>	#
<input checked="" type="checkbox"/> VoiceMail Main Menu <small>①</small>	*02
<input checked="" type="checkbox"/> Attended Transfer	*3
Attended Transfer Timeout <small>①</small>	15
<input checked="" type="checkbox"/> Blind Transfer	*03
<input checked="" type="checkbox"/> Call Pickup <small>①</small>	*4
<input checked="" type="checkbox"/> Extension Pickup <small>①</small>	*04
<input checked="" type="checkbox"/> Intercom <small>①</small>	*5
<input checked="" type="checkbox"/> Normal Spy <small>①</small>	*90
<input checked="" type="checkbox"/> Whisper Spy <small>①</small>	*91
<input checked="" type="checkbox"/> Barge Spy <small>①</small>	*92
Call Parking Preferences	
Call Parking	*6
Extension range used to park calls	690-699
Number of seconds a call can be parked for <small>①</small>	60
Call Forwarding Preferences	
<input checked="" type="checkbox"/> Reset to Defaults <small>①</small>	*70
<input checked="" type="checkbox"/> Enable Forward All Calls	*71
<input checked="" type="checkbox"/> Disable Forward All Calls	*071
<input checked="" type="checkbox"/> Enable Forward When Busy	*72
<input checked="" type="checkbox"/> Disable Forward When Busy	*072
<input checked="" type="checkbox"/> Enable Forward No Answer	*73
<input checked="" type="checkbox"/> Disable Forward No Answer	*073
<input checked="" type="checkbox"/> Forward to Number <small>①</small>	*74
<input checked="" type="checkbox"/> Forward to Voicemail	*074
<input checked="" type="checkbox"/> Enable Do Not Disturb	*75
<input checked="" type="checkbox"/> Disable Do Not Disturb	*075

Figure 6-46

1) General

•One Touch Record

A user may initiate or stop call recording by dialing *1 during a call. (*1 is default setting)

•Extension for Checking Voicemail

Users can check their Voicemail by dialing *2 on their phone (*2 is default setting).

•Voicemail for Extension

Users can leave a voicemail to other extensions by dialing # on their phone or

the incoming call could be forwarded to an extension's voicemail directly. (# is default setting).

For example, extension 500 want to leave a message for extension 501, users can use 500 dial '#501' to enter the voicemail of 501.

·Voicemail main menu

Users can go to the main menu by dialing *02 (*02 is default setting).

·Attended Transfer

Users may transfer an incoming call by dialing *3 on their phone (*3 is default setting).

·Attended Transfer Timeout

The time out of transferring a call

·Blind Transfer

Users may blind transfer an incoming call by dialing *03 on their phone (*03 is default setting).

·Call Pickup

Users may pick up an incoming call by dialing *4 on their phone (*4 is default setting)

·Extension Pickup

Users may pick up a specific extension's incoming call by dialing *04+extension number on their phone (*04 is default setting)

·Intercom

Define the feature code that is used to dial an extension in intercom mode. For instance setting this value to *5 would allow you to initiate an intercom call with extension 501 by dialing *5501.

·Normal Spy

In this mode, you can only listen to the extension being spied, for example you can dial *90501 to monitor extension 501

·Whisper Spy

In this mode you can listen/whisper to the extension being spied, for example, dialing *91501 to listen to extension 501, you can also talk with 501 too.

·Barge Spy

In this mode, you can barge in both extensions involved the call, for example dialing *92501 to barge in and talk with all the extensions inside

2) Call Park Preferences

·Call Parking

User may park an incoming call on his own telephone by pressing '*6'(*6 is default setting)

·Extension range used to park calls

User may park an incoming call on a designated extension at first and then pick up the call again on any other extension.

·Number of seconds a call can be parked before it is recalled.

Defines the number of seconds that a call can be parked before it is recalled to the station that parked it.

3) Call Forwarding Preferences

·Reset to Defaults

Users may reset all call forward defaults by calling *70 on their phone (*70 is default setting).

Note: When reset to defaults. The call forwarding settings will be configured as follows:

Always forward: Disabled

Busy forward to Voicemail: Enabled

No answer forward to Voicemail: Enabled

Do not disturb: Disabled

·Enable Forward All Calls

Users may enable always forward by calling *71 on their phone (*71 is default setting)

·Disable Forward All Calls

Users may disable always forward by calling *071 on their phone (*071 is default setting)

·Enable Forward When Busy

Users may enable busy forward by dialing *72 on their phone (*72 is default setting)

·Disable Forward When Busy

Users may disable busy forward by calling *072 on their phone (*072 is default setting)

·Enable Forward No Answer

Users may enable no answer forward by calling *73 on their phone (*73 is default setting)

·Disable Forward No Answer

Users may disable no answer forward by calling *073 on their phone (*072 is default setting)

·Forward to number

Users may activate call forwarding by dialing this feature code, followed by the extension or phone number to forward all calls to.

Note: Users may activate Forward to number by dialing *74 + phone number.
e.g.: by dialing *74501, all calls will be forwarded to extension 501.

·Forward to Voicemail

Users may forward the call to Voicemail by calling *074 on their phone (*074 is default setting)

·Enable Do Not Disturb

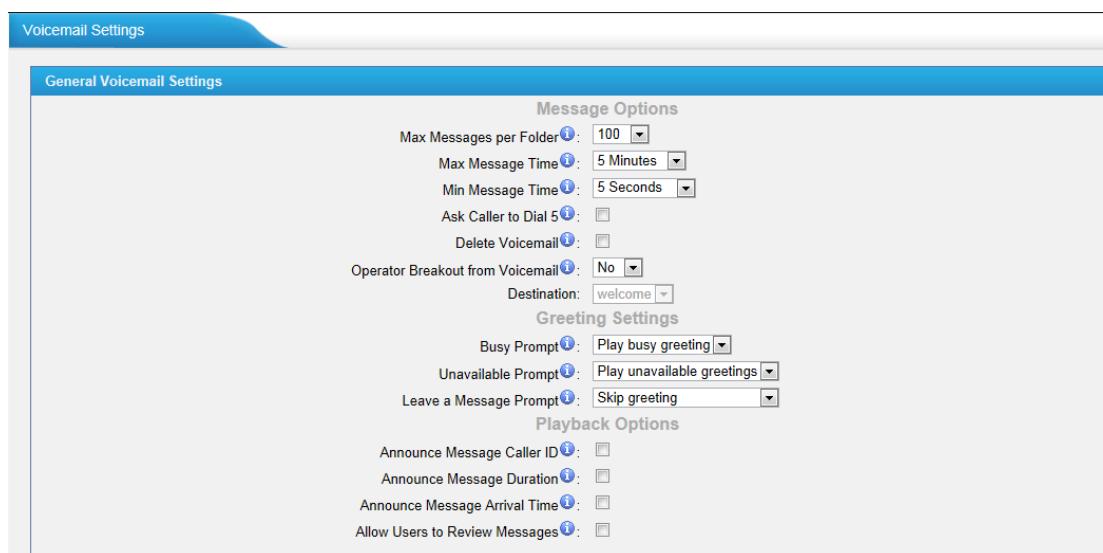
Users may enable do not disturb by calling *75 on their phone (*75 is default setting)

·Disable Do Not Disturb

Users may disable do not disturb by calling *075 on their phone (*075 is default setting)

6.6.4 Voicemail Settings

In this page, we can configure some settings for voicemail feature, including general voicemail settings and SMTP settings, which is used for 'voicemail to email'



The screenshot shows the 'General Voicemail Settings' page. It includes the following sections:

- Message Options**:
 - Max Messages per Folder: 100
 - Max Message Time: 5 Minutes
 - Min Message Time: 5 Seconds
 - Ask Caller to Dial 5:
 - Delete Voicemail:
 - Operator Breakout from Voicemail: No
 - Destination: welcome
- Greeting Settings**:
 - Busy Prompt: Play busy greeting
 - Unavailable Prompt: Play unavailable greetings
 - Leave a Message Prompt: Skip greeting
- Playback Options**:
 - Announce Message Caller ID:
 - Announce Message Duration:
 - Announce Message Arrival Time:
 - Allow Users to Review Messages:

Figure 6-47

1) General Voicemail Settings

a) Message Options

·Max Messages per Folder

Set the maximum number of messages that can be stored in a single voicemail box.

·Max Message Time

Set the maximum length of a single voicemail message.

·Min Message Time

Set the minimum length of a single voicemail message. Messages below this threshold will be automatically deleted.

·Ask Caller to Dial 5

If this option is set, the caller will be prompted to press 5 before leaving a message.

·Operator Breakout from Voicemail

If this option is set, the caller can jump out of the voicemail and go to the destination (IVR) you set by dialing "0".

b) Greeting Settings

·Busy Prompt

Greeting played when the extension called is busy.

Skip greeting: Do not play a greeting.

Play busy greeting: play the extension busy greeting.

·Unavailable Prompt

Greeting played when the extension called is Unavailable.

Skip greeting: Do not play a greeting.

Play Unavailable greeting: play the extension Unavailable greeting.

·Leave a Message Prompt

Greeting played to ask the caller to dial 5 to leave a message.

Skip greeting: Do not play a greeting.

Play busy greeting: play the extension busy greeting.

Play Unavailable greeting: play the extension Unavailable greeting.

c) Playback Options

·Announce Message Caller ID

If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

·Announce Message Duration

If this option is set, the duration of the message in minutes will be played back before the voicemail message begins playing.

.Announce Message Arrival Time

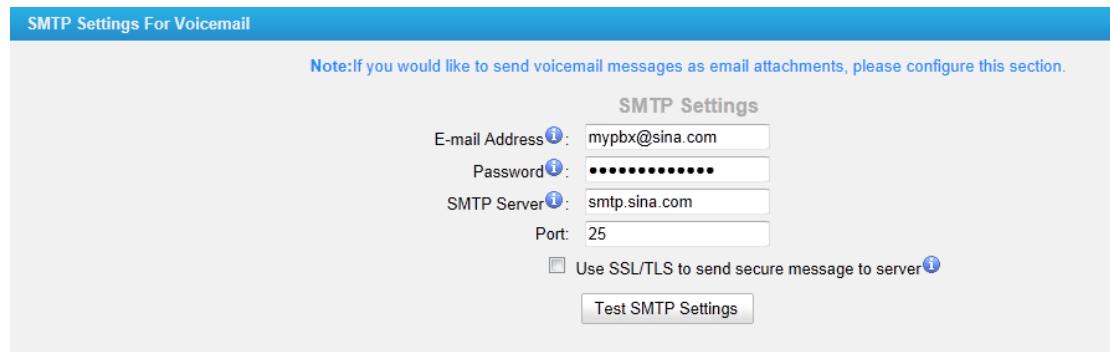
If this option is set, the arrival time of the message will be played back before the voicemail message begins playing.

.Allow Users to Review Messages

Allow callers to review their recorded message before sending it to voicemail.

2) SMTP Settings for Voicemail

Note: If you want to send voicemail messages as email attachments, please configure this section.



The screenshot shows the 'SMTP Settings For Voicemail' configuration page. At the top, a note says: 'Note:If you would like to send voicemail messages as email attachments, please configure this section.' Below this is a 'SMTP Settings' section with the following fields:

- E-mail Address: mypbx@sina.com
- Password: (redacted)
- SMTP Server: smtp.sina.com
- Port: 25
- Use SSL/TLS to send secure message to server (unchecked)

A 'Test SMTP Settings' button is located at the bottom of the form.

Figure 6-48

·E-mail Address

The E-mail Address that MyPBX will use to send voicemail.

·Password

The password for the email address used above

·SMTP Server

The IP address or hostname of an SMTP server that the MyPBX will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

·Port

SMTP Port: the default value is 25.

·Use SSL/TLS to send secure message to server

If the server of sending email needs to authenticate the sender, you need to select the check box.

Note: Must be selected for Gmail or exchange server.

After filling out the above information, you can click on the 'Test Account Settings' button to check whether the setup is OK.

1) If the test is successful, you can use the email safely.

2) If test failed, please check the above information is correct or network is proper.

6.7 Advanced Settings

6.7.1 SIP Settings

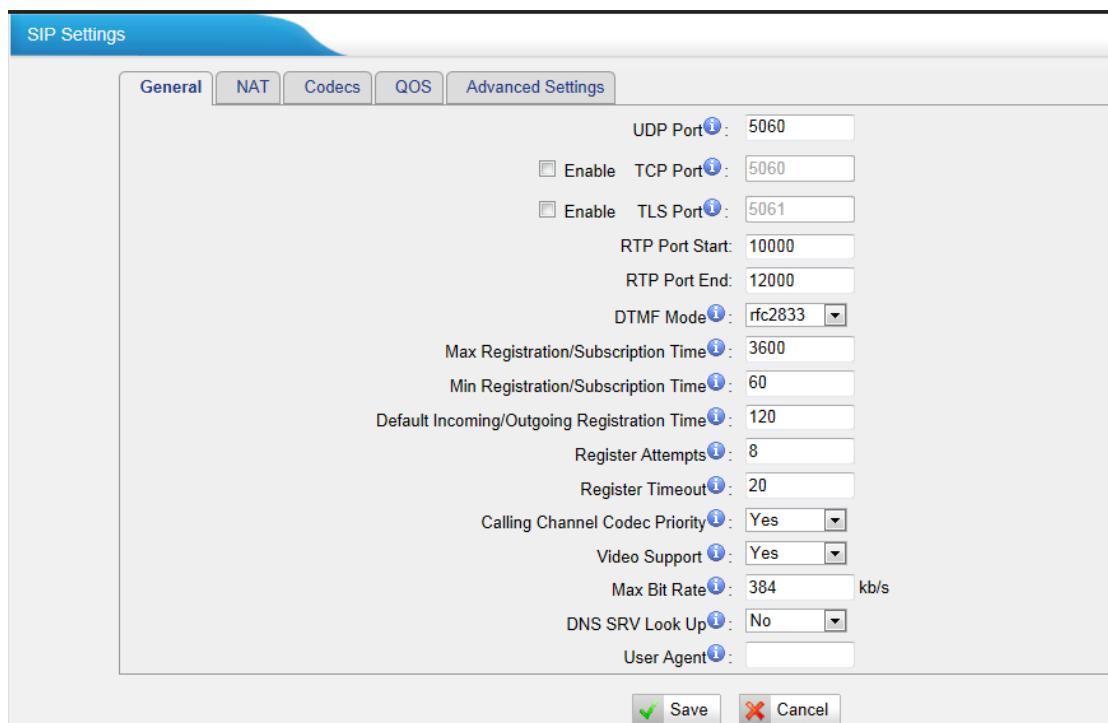


Figure 6-49

1) General

·UDP Port

Port use for sip registrations, Default is 5060.

·TCP Port

Port use for sip registrations, Default is 5060.

·TLS Port

Port use for sip registrations, Default is 5061.

·RTP Port Start

Beginning of RTP port range

·RTP Port End

End of RTP port range

·DTMF Mode

Set default mode for sending DTMF. Default setting: rfc2833

·Max Registration/Subscription Time

Maximum duration (in seconds) of a SIP registration. Default is 3600 seconds.

·Min Registration/Subscription Time

Minimum duration (in seconds) of a SIP registration. Default is 60seconds.

·Default Incoming/Outgoing Registration Time

Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.

·Register Attempts

The number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default is 8 times.

·Register Timeout

Number of seconds to wait for a response from a SIP Registrar before timed out . Default is 20 seconds.

·Calling Channel Codec Priority

Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected in preference. If not, MyPBX will follow the priority in your SIP/SPS trunks.

·Video Support

Support for SIP video or no. Default is yes.

·Max Bit Rate

Configure the max bit rate for video stream. The default: 384kb/s

·DNS SRV Look Up

Please enable this option when your SIP trunk contains more than one IP address.

·User Agent

To change the user agent parameter of asterisk, the default is 'MyPBX', you should change it if needed.

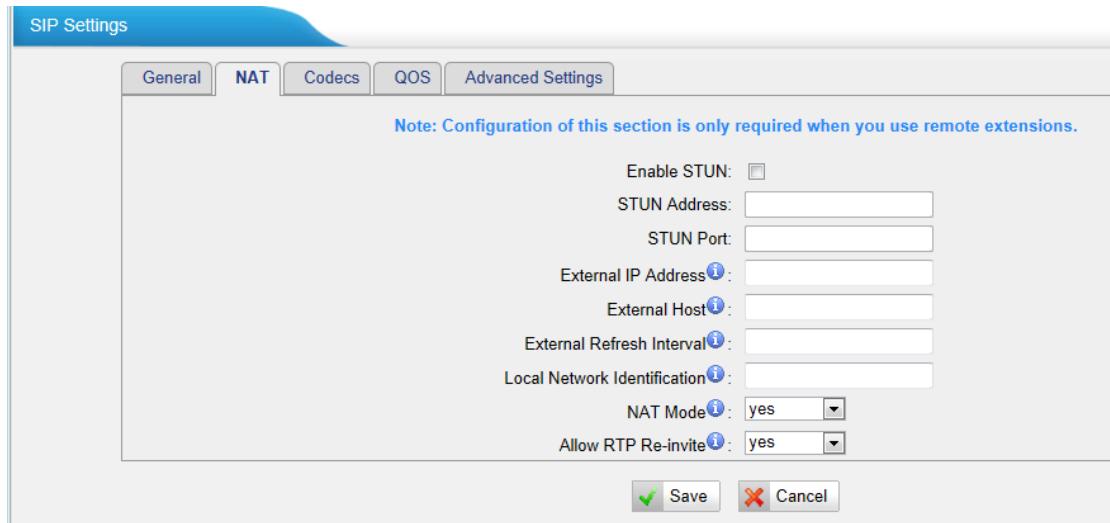


Figure 6-50

2) NAT

Note: Configuration of this section is only required when using remote extensions.

·Enable STUN

STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

·STUN Address

The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.

·External IP Address

The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.

·External Host

Alternatively you can specify an external host, and the system will perform DNS queries periodically.

This setting is only required when your public IP address is not static. It is recommended that a static public IP address be used with this system. Please contact your ISP for more information.

·External Refresh Interval

If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.

·Local Network Identification

Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall.

Some examples of this are as follows:

'192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks;

'10.0.0.0/255.0.0.0' : Also RFC1918;

'172.16.0.0/12':AnotherRFC1918withCIDRnotation;

'169.254.0.0/255.255.0.0' : Zero conf local network.

Please refer to RFC1918 for more information.

•NAT Mode

Global NAT configuration for the system. The options for this setting are as follows:

Yes = Use NAT. Ignore address information in the SIP/SDP headers and reply to the sender's IP address/port.

No = Use NAT mode only according to RFC3581.

Never = Never attempt NAT mode or RFC3581 support.

Route = Use NAT but do not include report in headers.

•Allow RTP Reinvite

By default, the system will route media streams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.

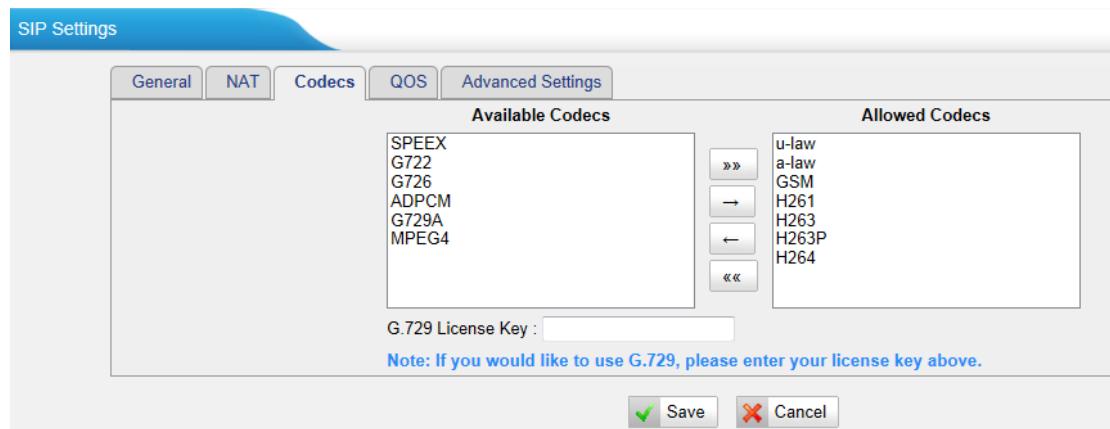


Figure 6-51

3) Codecs

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.

u-law: A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

a-law: A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

GSM: A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

SPEEX: Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

G.722: G.722 is a wideband speech coding algorithms which supports the bit rate of 64, 56 and 48kbps wideband. It's a broadband voice encoding of G series.

G.726: A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

ADPCM, G.729A, H261, H263, H263p, H264, MPEG4.

Note: If you would like to use G.729, please enter your license.

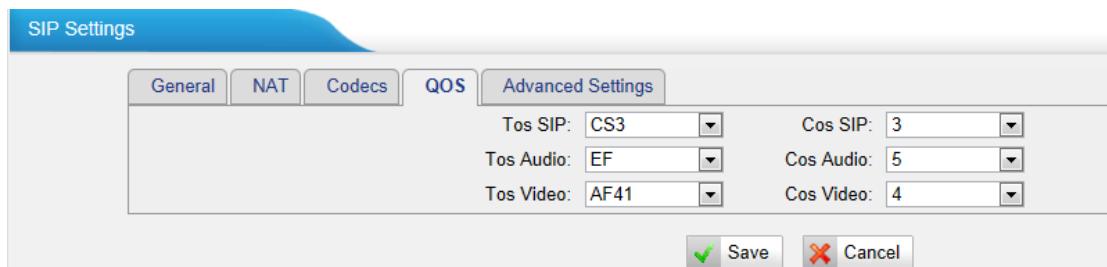


Figure 6-52

4) QoS

QoS (Quality of Service) is a major issue in VOIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic. When the network capacity is insufficient, QoS could provide priority to users by setting the value.

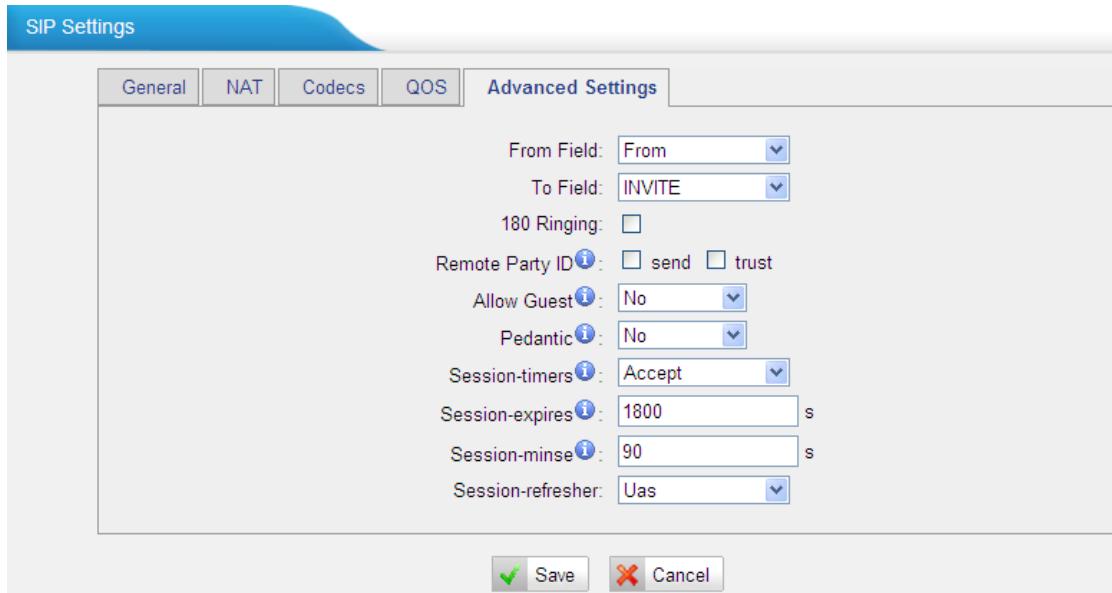


Figure 6-53

5) Advanced Settings

·From Field

Where to get the caller ID in sip packet.

·To Field

Where to get the DID in sip packet.

·180 Ringing

It is set when the telecom provider needs. Usually it is not needed.

·Qualify

Send check alive packets to the sip provider.

·Remote Party ID

Whether send Remote-Party-ID on SIP header. Default no.

·Allow Guest

Whether allow anonymous registration extension. Default: no.

This option is used to avoid some anonymous calls by hackers. More details for the system security configuration, please refer to **APPENDIX G MyPBX**

Security Configuration Guide.

·Pedantic

Enable pedantic parameter. Default: no.

·Session -timers

Enable sesstion-timer mode, default: yes

·Session-expire

The max refresh interval

·Session-minse

The min refresh interval, which mustn't be less than 90s

·Session-refresher

Choose session-refresher, the default is Uas

6.7.2 IAX Settings

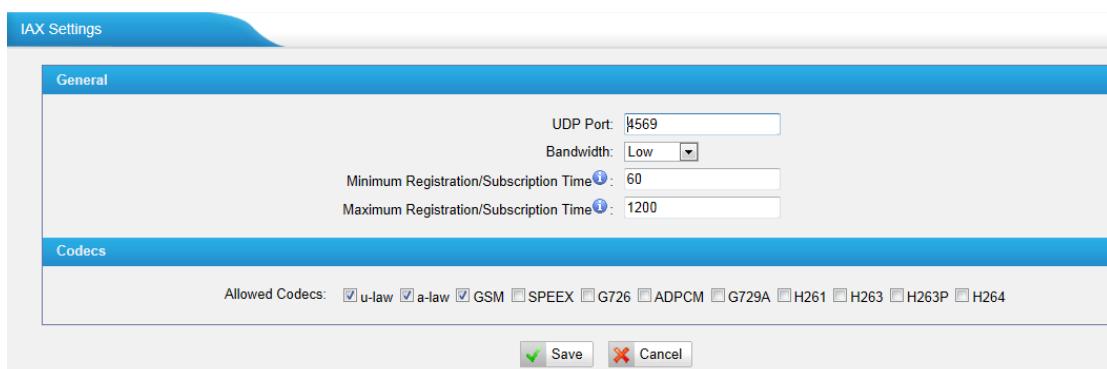


Figure 6-54

1) General

·Bind Port

Port use for IAX2 registrations, Default is 4569.

·Bandwidth

Low/medium/high with this option you can control which codec to be used.

·Min Registration Time

Minimum duration (in seconds) of a IAX2 registration. Default is 60seconds.

·Max Registration Time

Maximum duration (in seconds) of a IAX2 registration. Default is 1200 seconds.

2) Codecs

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.

u-law: A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

a-law: A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

GSM: A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

SPEEX: Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

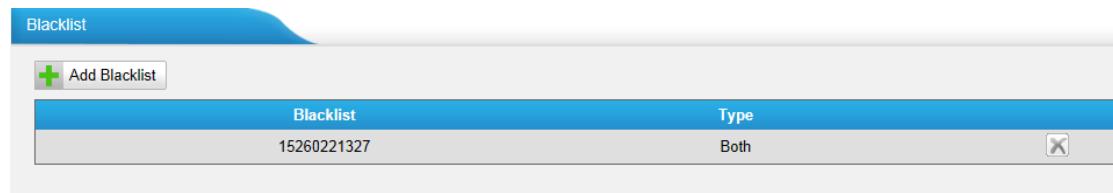
G.726: A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

ADPCM, G.729A, H261, H263, H263p, H264.

Note: If you would like to use G.729, please enter your license.

6.7.3 Blacklist

Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is registered in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.



We can add a number with the type: inbound, outbound or both

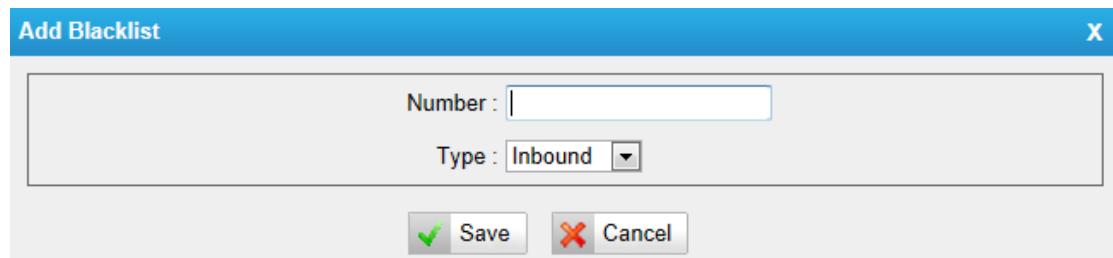


Figure 6-55

Note: It will not working for internal extension number.

6.7.4 Callback Settings

MyPBX allows caller A to dial an inbound route number, and after hearing the ring, A can hang up the call or wait for MyPBX to cut off the call, then MyPBX will call A with this number. When A picks up the call, A can dial the number he wants to call; MyPBX will call the number with its outbound route.

Note:

1. If you'd like to use callback feature, please make sure if it's enabled on the inbound route setting panel.
2. No callback rules needed to be set if the trunk supports call back with the caller ID directly.

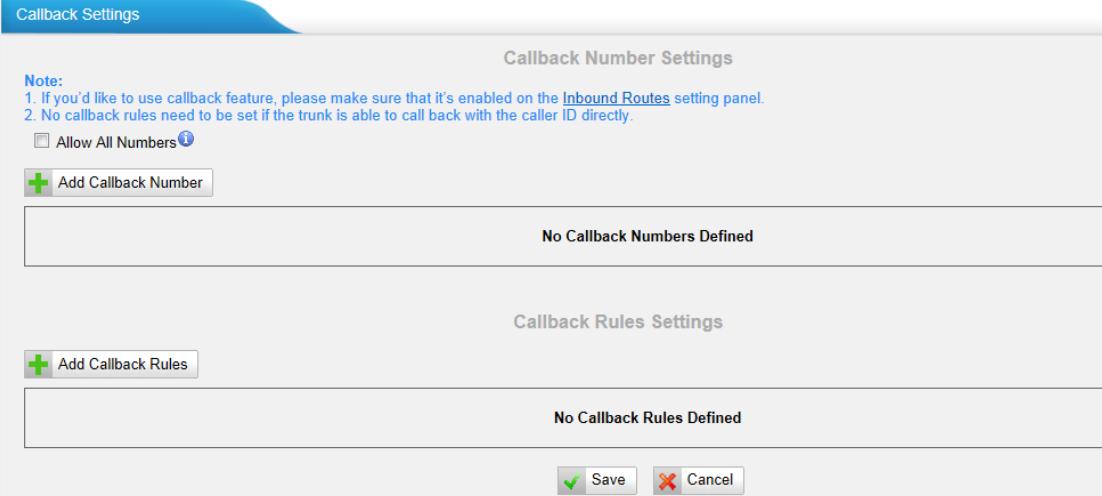


Figure 6-56

•Allow All Numbers

If you want to apply Callback function to all incoming numbers, please tick Allow All numbers.

Follow the step to use this function.

Step 1: Enable Callback.

Inbound Routes –Choose “Yes” on “Enable Callback” to enable this function

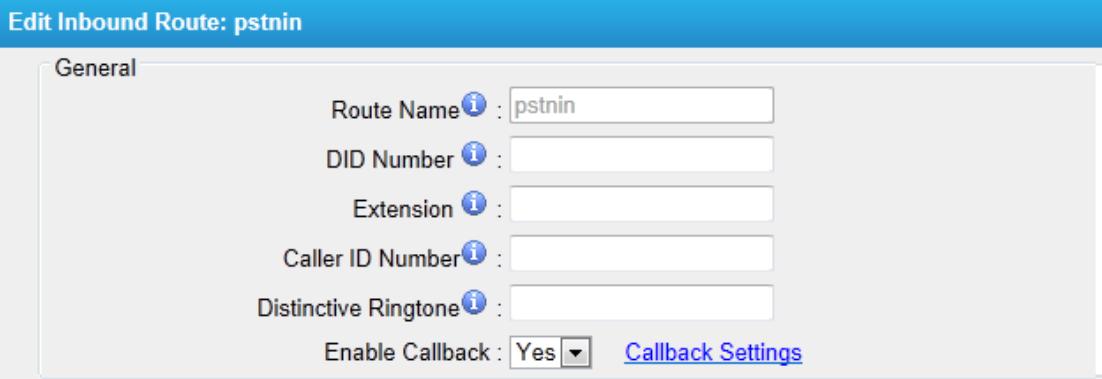


Figure 6-57

Step 2: Create Callback number

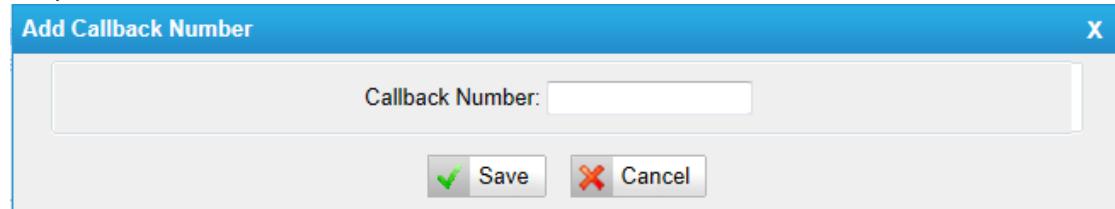


Figure 6-58

Step 3: Create Callback Rules

You will need to create callback rules when the system should strip or add digits.

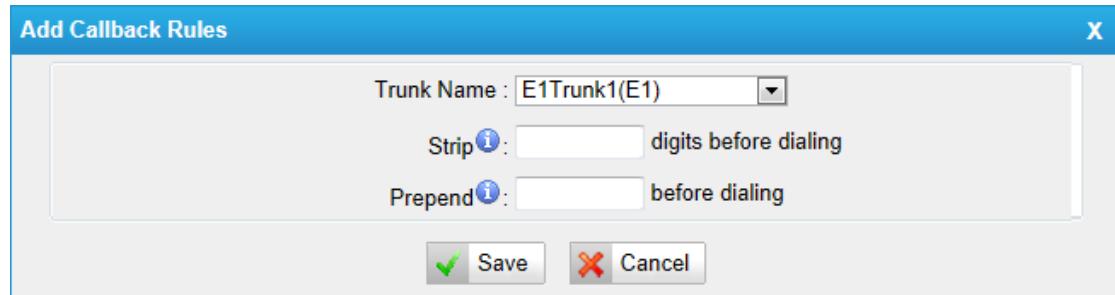


Figure 6-59

·Trunk Name

Choose the trunk with callback rules

·Strip digits from front

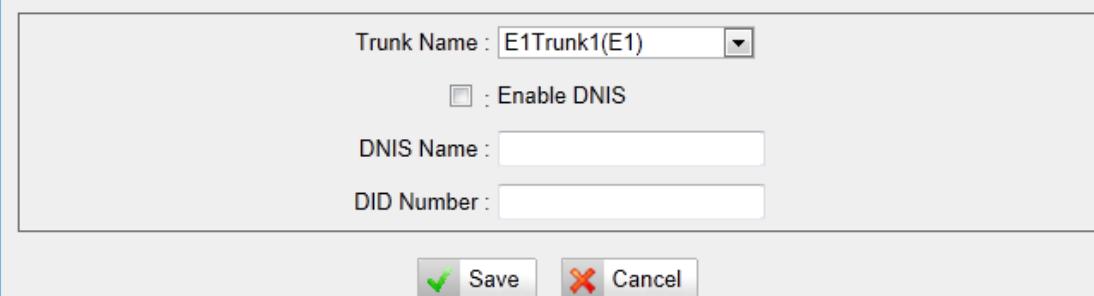
Define how many digits will be stripped from the call in number before the callback is placed. For example, when you call from number 123456789 into MyPBX, the caller ID is 0123456789, but you can only call 123456789 successfully from MyPBX trunk. You should configure number 0123456789 as the call back number and strip 1 digit before the callback is placed

·Prepend before dialing

Define digits added before a callback number before the callback is placed. For example, the call in number (Caller ID) is 123456789, MyPBX need to send 9123456789 to its trunk when call to this number. You should configure 123456789 as the call back number and add 9 before the callback is placed. You can add 'w' for analog trunks for some delay too

6.7.5 DNIS Settings

DNIS (Dialed Number Identification Service) is a telephone service that identifies for the receiver of a call the number that the caller dialed



The screenshot shows a configuration dialog titled 'Add DNIS'. It contains the following fields:

- Trunk Name : E1Trunk1(E1)
- : Enable DNIS
- DNIS Name : [empty input field]
- DID Number : [empty input field]

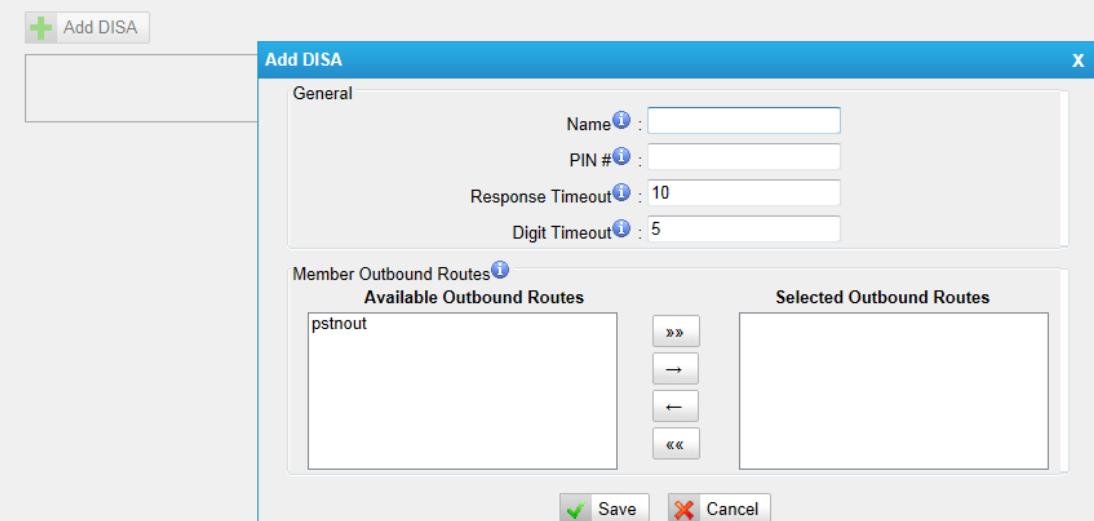
At the bottom are two buttons: a green checkmark labeled 'Save' and a red X labeled 'Cancel'.

Figure 6-60

Note: If DID is not configured here, all the calls via this trunk will show the DNIS instead of the original caller ID

6.7.6 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an 'internal' system dial tone and make calls as if they were using one of the extensions attached to the telephone switch. To use DISA, a user calls a DISA number, which invokes the DISA application. The DISA application in turn requires the user to enter a PIN number, followed by the pound sign (#). If the PIN number is correct, the user will hear dial tone on which a call may be placed. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security



The screenshot shows a configuration dialog titled 'Add DISA'. It has two main sections:

- General** section:
 - Name : [empty input field]
 - PIN # : [empty input field]
 - Response Timeout : 10
 - Digit Timeout : 5
- Member Outbound Routes** section:
 - Available Outbound Routes**: A list containing 'pstnout'.
 - Selected Outbound Routes**: An empty list.
 - Buttons between the lists: '»»', '→', '←', and '««'.

At the bottom are two buttons: a green checkmark labeled 'Save' and a red X labeled 'Cancel'.

Figure 6-61

1) General

·DISA Name

Give this DISA application a name to help you identify it.

·PIN#

The password for this DISA .

·Response Timeout

The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. Default is 10 seconds.

·Digit Timeout

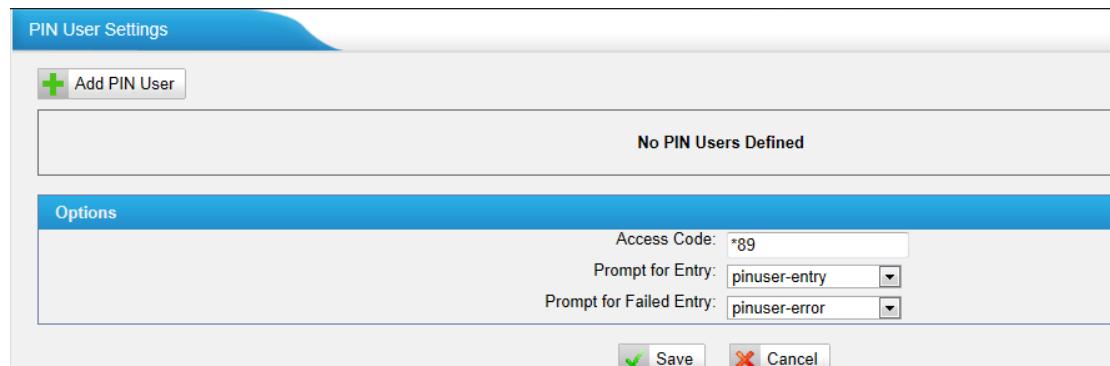
The maximum amount of time permitted between each digit when the user is dialing an extension number. Default is 5 seconds.

2) Member Outbound Routes

Used to set the outbound routes that can be accessed from this DISA.

6.7.5 PIN User Settings

PIN User is used to manage lists of PINs that can be used to access restricted features such as Outbound Routes.



The screenshot shows a software interface titled "PIN User Settings". At the top, there is a button labeled "Add PIN User". Below this, a message says "No PIN Users Defined". Under the heading "Options", there are three input fields: "Access Code" (set to "*89"), "Prompt for Entry" (set to "pinuser-entry"), and "Prompt for Failed Entry" (set to "pinuser-error"). At the bottom right are two buttons: "Save" (with a checkmark icon) and "Cancel".

Figure 6-62

1) Options

·Access Code

.Dial this code to access PIN.

·Prompt for Entry

Prompt caller enter the PIN Number.

·Prompt for Entry Failure

Prompt the caller when an invalid PIN is entered.

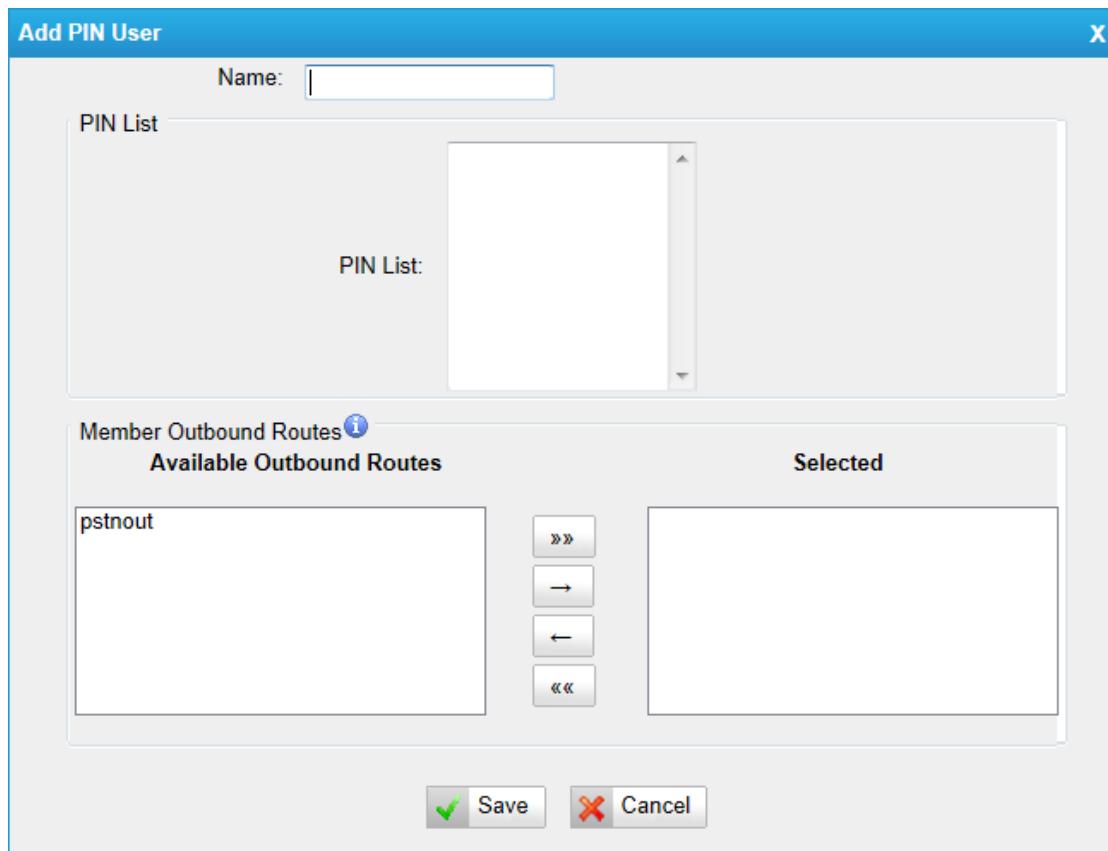


Figure 6-63

2) PIN User

MyPBX can store a number of PIN Users. PIN Users may be used to keep track of calls in relation to particular activities or clients. They can also be used to keep track of calls by particular users or sets of users.

- PIN entered are checked against those stored by the system. If an invalid PIN is entered, the PIN is requested again.
- The system administrator can configure certain numbers or types of numbers to require entry of a PIN before you can continue making a call to such a number.
- The system administrator can also configure you to have to enter a PIN before making any external call.

·Name

A character-based name for this PIN list, i.e. 'Yeastar PIN'

·PIN List

Enter a list of one or more PINs, One PIN per line.

·Outbound Route

PIN User can use those outbound route to make call out

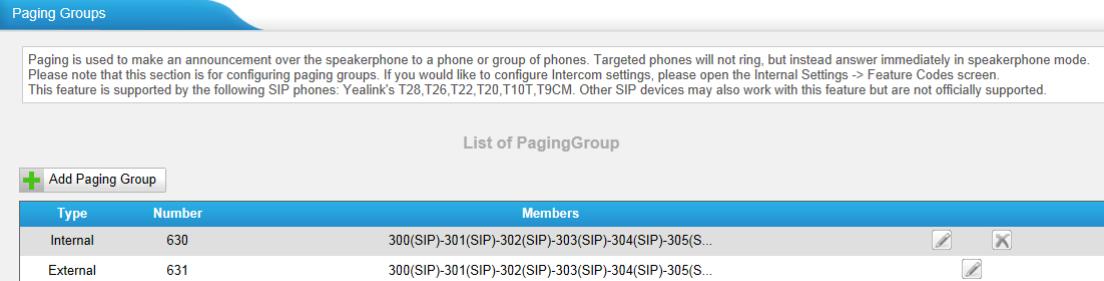
6.7.8 Paging Groups

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the Other Settings -> Feature Codes screen.

This feature is supported by the following SIP phones:

Yealink's T28, T26, T22, T20, T10T, T9CM. Other SIP devices may also work with this feature but are not officially supported.

Note: A paging group can have a maximum of 20 members.



Type	Number	Members
Internal	630	300(SIP)-301(SIP)-302(SIP)-303(SIP)-304(SIP)-305(SIP)
External	631	300(SIP)-301(SIP)-302(SIP)-303(SIP)-304(SIP)-305(SIP)

Figure 6-64

There are two types of paging groups in MyPBX U300

1. Internal paging Group

In this mode, if you dial its number, MyPBX will help to pick up those chosen members and you can talk directly without any rings.

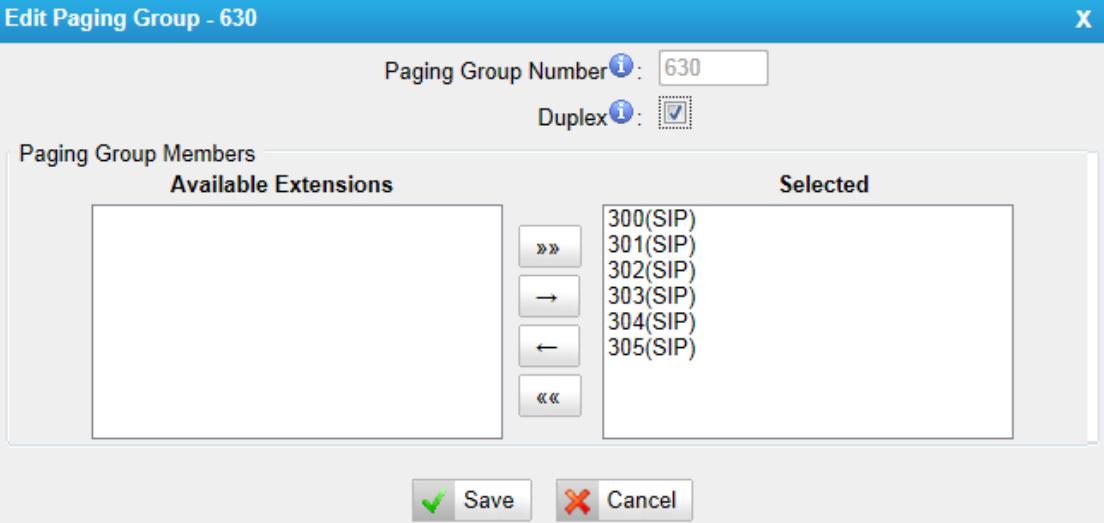


Figure 6-65

Paging Group Number

Define the numbered extension that may be dialed to reach this group.

Duplex

Paging is typically one way for announcements only. Checking this will make paging duplex, allowing all users in the group to talk and be heard by all.

2. External paging group

In this mode, the chosen extensions will have the rights to dial the group number, all his voice will be broadcasted via the 'Audio Out' interface of U300

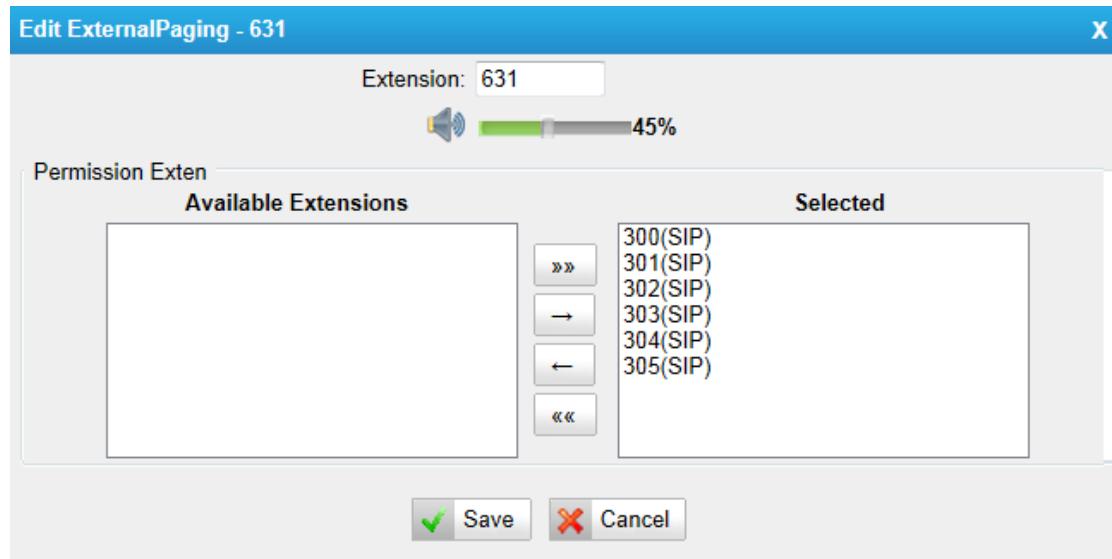


Figure 6-66

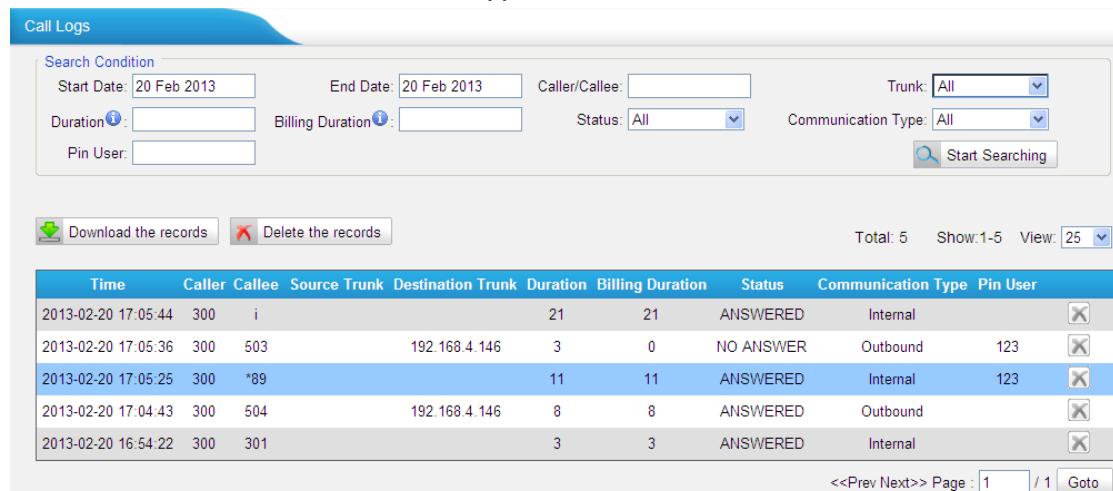
7 Reports


Click **Reports** to access

We can check the call detailed logs for counting and system log for debugging

7.1 Call Logs

The call Log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by filter the call logs by call date, caller/callee, trunk, duration, billing duration, status, communication type, Pin User.



Call Logs

Search Condition

Start Date: 20 Feb 2013	End Date: 20 Feb 2013	Caller/Callee:	Trunk: All
Duration: <input type="text"/>	Billing Duration: <input type="text"/>	Status: All	Communication Type: All
Pin User: <input type="text"/>	<input type="button" value="Start Searching"/>		

Action Buttons: Download the records | Delete the records

Total: 5 Show: 1-5 View: 25

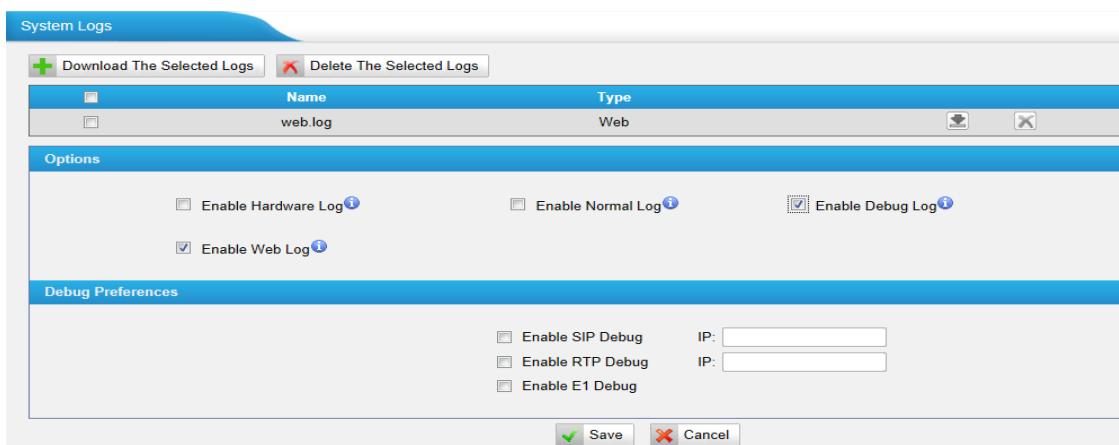
Time	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Type	Pin User
2013-02-20 17:05:44	300	i			21	21	ANSWERED	Internal	<input checked="" type="checkbox"/>
2013-02-20 17:05:36	300	503		192.168.4.146	3	0	NO ANSWER	Outbound	123 <input checked="" type="checkbox"/>
2013-02-20 17:05:25	300	*89			11	11	ANSWERED	Internal	123 <input checked="" type="checkbox"/>
2013-02-20 17:04:43	300	504		192.168.4.146	8	8	ANSWERED	Outbound	<input checked="" type="checkbox"/>
2013-02-20 16:54:22	300	301			3	3	ANSWERED	Internal	<input checked="" type="checkbox"/>

<<Prev Next>> Page: 1 / 1 Goto

Figure 7-1

7.2 System Logs

You can download and delete the system logs of MyPBX



System Logs

Action Buttons: Download The Selected Logs | Delete The Selected Logs

Name	Type
web.log	Web

Options

- Enable Hardware Log ①
- Enable Normal Log ①
- Enable Debug Log ①
- Enable Web Log ①

Debug Preferences

<input type="checkbox"/> Enable SIP Debug	IP: <input type="text"/>
<input type="checkbox"/> Enable RTP Debug	IP: <input type="text"/>
<input type="checkbox"/> Enable E1 Debug	

Figure 7-2

Options

·Enable Hardware Log

Save the information of hardware; (up to 4 log files)

·Enable Normal Log

Save the prompt information; (up to 16 log files)

·Enable Web Log

Save the history of web operations (up to 2 log files)

·Enable Debug Log

Save debug information (up to 2 log files)

We can also enable the SIP/RTP/E1 debug, it's very useful for checking problems.

2) Packet Capture Tool

This feature always used to capture packets for technician. Integrate packet capture tool 'Wireshark' in MyPBX.

Users also could specify the destination IP address and port to get the packets.

·IP

Specify the destination IP address to get the packets.

·Port

Specify the destination Port to get the packets.

8 Logout



Click **Logout** to log out safely to the log in page.

9. Use MyPBX

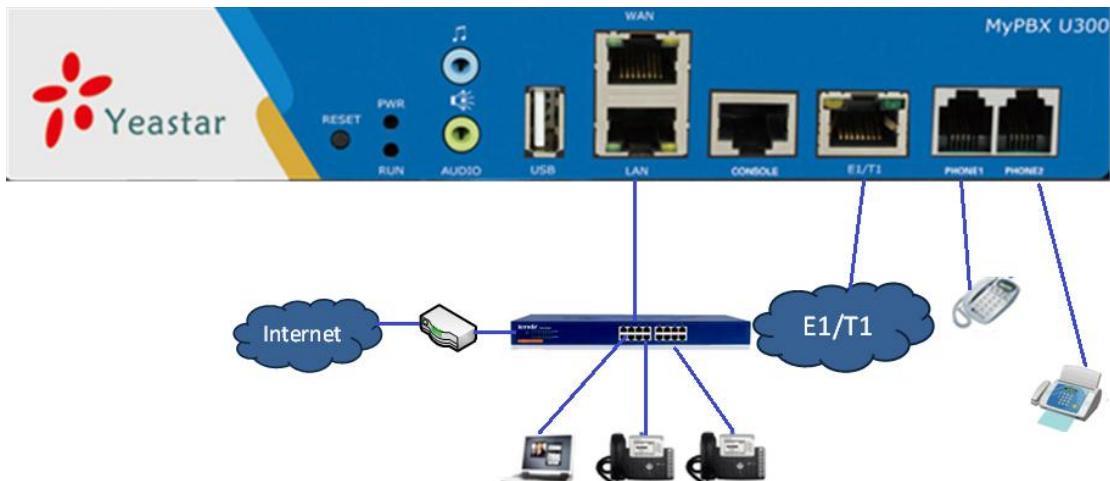


Figure 9-1

9.1 Make outbound call

To make an outbound call, we need to add trunk first. There are 3 types of VoIP Trunk:

- Digital Trunk:** E1/T/J1 ports of MyPBX, connected to the ISDN provider.
You need to get the detailed signaling type from the provider before configure the E1 trunk.
- VoIP Trunk:** Connected to remote VOIP service server.
You should get an IP address with user name/ password from the provider.
- Service Provider:** Connected to service provider server.
You will get only IP address for authorization.

9.1.1 Sample Routing via E1 Trunk

Let's route all inside extensions through an E1 trunk by dialing 9. In Outbound Routes, add a new outbound route as below.

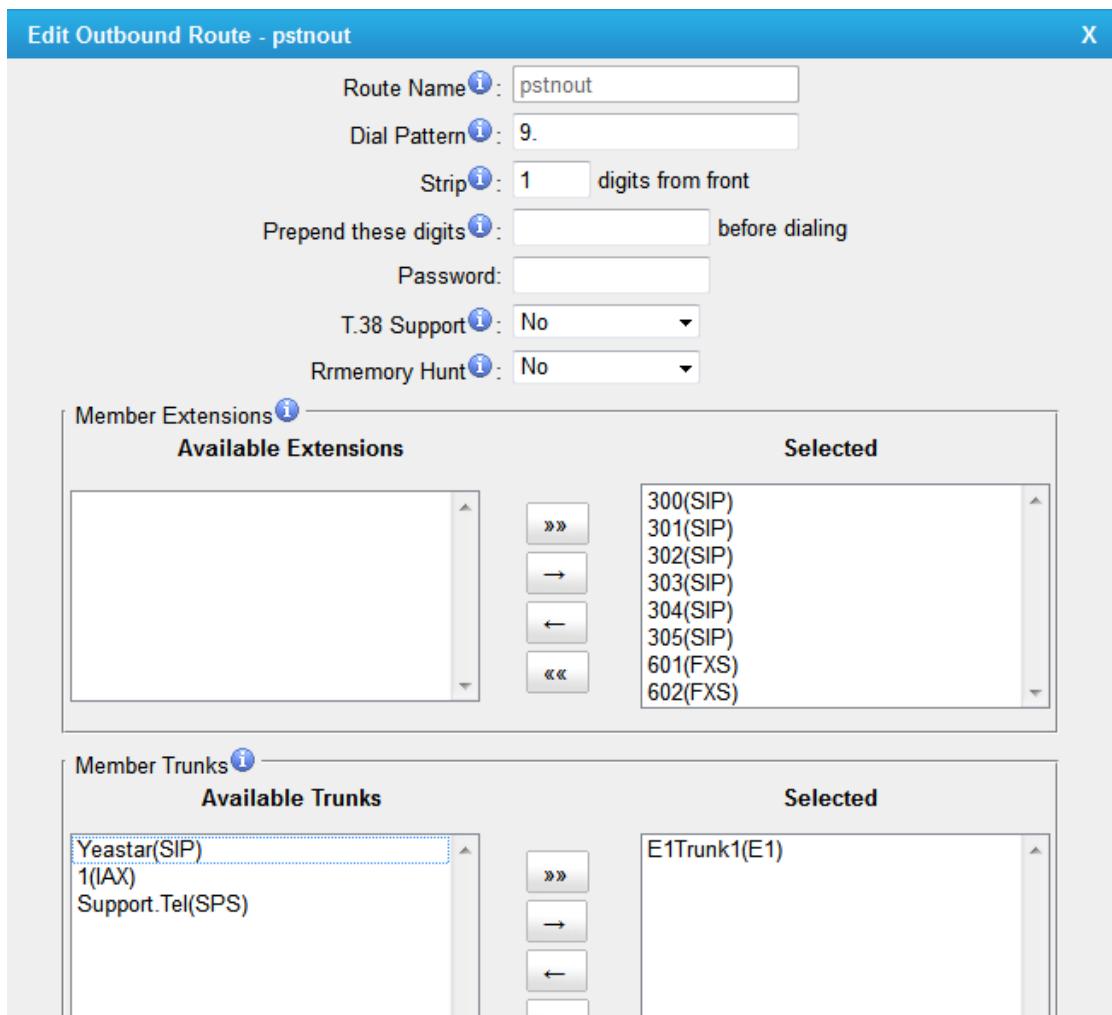


Figure 9-1

As we can see from the outbound route of 'pstnout', all phone numbers starting with 9 will have their first digit stripped off (digit 9) and will be sent to the E1.

After we have configured the above, we can dial 9 + local number to dial out via a E1 trunk.

Note: Setting number prefix to wild card **X**, and setting Strip to **0** digits from the front will allow all calls to go through this outbound route directly.

9.1.2 Sample Routing via VoIP Trunk

Let's configure all inside extensions to dial '0' through the VoIP Trunk.

1. Add VoIP service provider

Before we do add this, please make sure you have a VoIP Trunk account.
Trunks→VoIP Trunk →SIP Trunk

Enter your account information on this page, and click Save.

Created New VoIP trunk

Type:	SIP	
Provider Name:	voipprovider	
Hostname/IP:	voip.6699.org	: 5060
Domain:	voip.6699.org	
User Name:	16885885	
Authorization Name:	16885885	
Password:	*****	
From User:	16885885	
Online Number <small>i</small> :		
Maximum Channels <small>i</small> :	1	
<input type="checkbox"/> Enable Outbound Proxy Server		
Transport:	UDP	Enable SRTP <small>i</small> : <input type="checkbox"/>
Caller ID <small>i</small> :		
DOD Setting		
DOD:	Associated Extension: 500	<input type="button" value="↑Add DOD"/>
<input type="button" value="Save"/> <input type="button" value="Cancel"/>		

Figure9-2

2. Add Outbound Routes

As we can see from the Outbound Route of 'voipout', all phone numbers starting with 0 will have their first digit stripped off (digit 0) and will be sent to the SIP Trunk.

New Outbound Route

Route Name <small>i</small> :	vopayout
Dial Pattern <small>i</small> :	0.
Strip <small>i</small> :	1 digits from front
Prepend these digits <small>i</small> :	
Password:	

Member Extensions i

Available Extensions	Selected
	500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP)

Member Trunks i

Available Trunks	Selected
	voipprovider(SIP) 8032(SIP)

Save
 Cancel

Figure9-3

Now that we have added two outbound dialing rules, any call starting with 9 will be routed to the PSTN, and any number starting with 0 will be routed to the SIP Trunk.

9.2 Incoming call

9.2.1 Sample Routing to an IVR

Let's configure an incoming call to route to the IVR. In the IVR itself, let's configure digit 0 to route the call to extension 500, and digit 1 to route the call to extension 501.

1. Add IVR

To add a new IVR, go to IVR → Create New IVR

Edit IVR welcome

Number i :	660																																														
Name i :	welcome																																														
Prompt i :	default ▼	Custom IVR Prompts																																													
Play times i :	3 ▼																																														
WaitExten i :	3 ▼																																														
<input checked="" type="checkbox"/> i Allow Dialing Other Extensions																																															
KeyPress Events <table border="1"> <thead> <tr> <th>Key</th> <th>Action</th> <th>Destination</th> </tr> </thead> <tbody> <tr><td>0</td><td>Connect to Extension ▼</td><td>User Extension -- 500 ▼</td></tr> <tr><td>1</td><td>Connect to Extension ▼</td><td>User Extension -- 501 ▼</td></tr> <tr><td>2</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>3</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>4</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>5</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>6</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>7</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>8</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>9</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>#</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>*</td><td>No Action ▼</td><td>▼</td></tr> <tr><td>TimeOut i</td><td>Connect to Extension ▼</td><td>User Extension -- 500 ▼</td></tr> <tr><td>Invalid i</td><td>Connect to Extension ▼</td><td>User Extension -- 500 ▼</td></tr> </tbody> </table>			Key	Action	Destination	0	Connect to Extension ▼	User Extension -- 500 ▼	1	Connect to Extension ▼	User Extension -- 501 ▼	2	No Action ▼	▼	3	No Action ▼	▼	4	No Action ▼	▼	5	No Action ▼	▼	6	No Action ▼	▼	7	No Action ▼	▼	8	No Action ▼	▼	9	No Action ▼	▼	#	No Action ▼	▼	*	No Action ▼	▼	TimeOut i	Connect to Extension ▼	User Extension -- 500 ▼	Invalid i	Connect to Extension ▼	User Extension -- 500 ▼
Key	Action	Destination																																													
0	Connect to Extension ▼	User Extension -- 500 ▼																																													
1	Connect to Extension ▼	User Extension -- 501 ▼																																													
2	No Action ▼	▼																																													
3	No Action ▼	▼																																													
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9	No Action ▼	▼																																													
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*	No Action ▼	▼																																													
TimeOut i	Connect to Extension ▼	User Extension -- 500 ▼																																													
Invalid i	Connect to Extension ▼	User Extension -- 500 ▼																																													
<input checked="" type="checkbox"/> Save <input type="button" value="Cancel"/>																																															

Figure9-4

2. Add Inbound Routes

As we can see from the Inbound Route of 'allin', all incoming calls will be sent to the IVR.

create New Inbound Route

General

Route Name <small>i</small>	allin
DID Number <small>i</small>	
Extension <small>i</small>	
Caller ID Number <small>i</small>	
Distinctive Ringtone <small>i</small>	

Member Trunksi

Available Trunks	Selected
[Empty]	8032(SIP) voipprovider(SIP)

During Office Hours

<input type="radio"/> End Call	
<input type="radio"/> Extension	Extension -- 500
<input type="radio"/> Voicemail	Voicemail -- 500
Destination:	<input checked="" type="radio"/> IVR
	IVR -- welcome
	<input type="radio"/> RingGroup
	RingGroup -- ringgroup_defi
	<input type="radio"/> Conference Room
	Conference Room -- 640
	<input type="radio"/> DISA
	DISA --
	<input type="radio"/> Queues
	Queues --
	<input type="radio"/> Faxes <small>i</small>
	Faxes -- 500
	<input type="radio"/> Outbound Routes <small>i</small>
	Route Name -- pstnout

Outside Office Hours

<input type="radio"/> End Call	
<input type="radio"/> Extension	Extension -- 500
<input type="radio"/> Voicemail	Voicemail -- 500
Destination:	<input checked="" type="radio"/> IVR
	IVR -- welcome
	<input type="radio"/> RingGroup
	RingGroup -- ringgroup_defi
	<input type="radio"/> Conference Room
	Conference Room -- 640
	<input type="radio"/> DISA
	DISA --
	<input type="radio"/> Queues
	Queues --
	<input type="radio"/> Faxes <small>i</small>
	Faxes -- 500
	<input type="radio"/> Outbound Routes <small>i</small>
	Route Name -- pstnout

Action Buttons

Save

Figure9-5

APPENDIX A FAQ

Q1. How to Register SIP device?

A1:

1) Register SIP softphone

Download the x-lite softphone from counterpath website

www.counterpath.com

After install the x-lite, right click the panel and select the SIP Account setting and then configure it.

Display Name: 500

User Name: 500

Password: 500

Authorization Name: 500

Domain: 192.168.5.150

2) Register IP Phone (for example, Yealink's T28 IP Phone)

a) Connect the T28's Internet port to the switch. And it can get the IP from your route.

b) Press the 'OK' key on T28 to get the IP of T28.

c) Put the IP on web browser then you can enter the T28 configure page through this IP.

d) Put the SIP extensions info on the T28 IP phones.

Display Name: 501

User Name: 501

Register Name: 501

Password: 501

SIP Server: 192.168.5.150

Use the same method register another T28 to other extension.

Q2. How do I reset MyPBX back to the factory default settings?

A2: To perform a reset, please follow steps below:

Step 1: Hold down the 'Reset' button on the back of the unit for 5 seconds and watch the LEDs on the front of the MyPBX. When the status LED turns red, let go of the reset button.

Step 2: When the RUN status LED starts blinking, MyPBX will be set back to factory defaults.

Step 3: To access the configuration page, navigate to **192.168.5.150** using a web browser. Make sure that you are on the 192.168.5.0 subnet before doing this.

Step 4: Login to the device with the username '**admin**' and the password '**password**', in order to begin reconfiguring the device.

APPENDIX B How to Configure Autobackup

Before Autobackup can be properly configured, an SMB share folder accessible from MyPBX must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

Step 1 Add a new folder, rename it, and set this new folder's share Properties according to Figure B-1

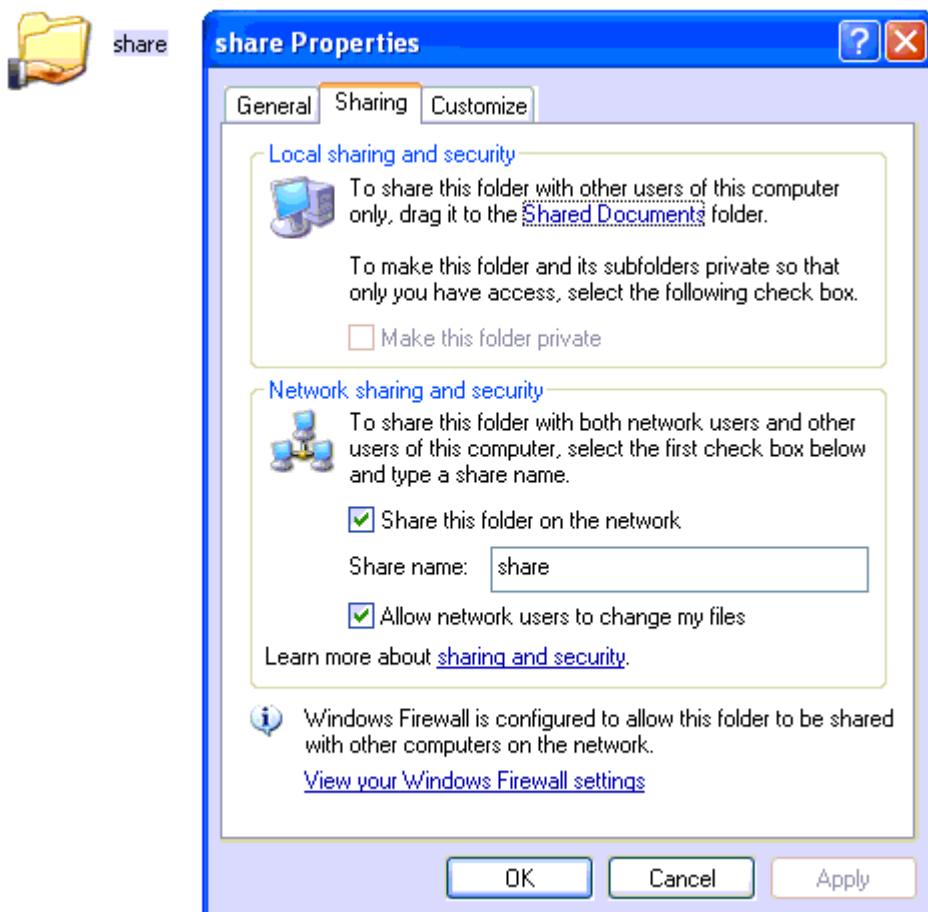


Figure B-1 Set up share Properties

Step 2 Enter the new folder and create a new text file, then rename this file to status.txt. This step is very important, DO NOT forget to create the status.txt file.

Step 3 Configure Autobackup settings on MyPBX to Figure B-2

External Storage Settings

The External Storage feature is used to extend storage space. Once configured, the files(voicemail, call recording files, CDR files) created before the configured days will be moved to the Net-Disk.

<u>Step 1: Create a Net-Disk on a chosen computer</u>		
<u>Step 2: Input the Net-Disk properties</u>		
Net-Disk Host/IP:	<input type="text" value="192.168.5.222"/>	
Net-Disk Share Name:	<input type="text" value="share"/>	
Net-Disk Access Username:	<input type="text"/>	
Net-Disk Access Password:	<input type="password"/>	
Move files created before:	<input type="button" value="1"/> <input type="button" value="▼"/> days ago	<input type="button" value="Save"/> <input type="button" value="Cancel"/>
<u>Step 3: Save Net-Disk settings</u>		
<u>Step 4: Make sure the settings are successfully completed</u>		

Figure B-2 Autobackup Setting

Net-Disk Host/IP: Change this to the IP address of the computer where backup files will be stored.

Net-DiskShare Name: Change this to the name of the shared folder where backups will be stored.

Net-DiskShare Username: The user name used to log into the network share. Leave this blank if it is not required

Net-DiskShare Password: The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to Figure B-3, then you have successfully configured Autobackup on the MyPBX unit.

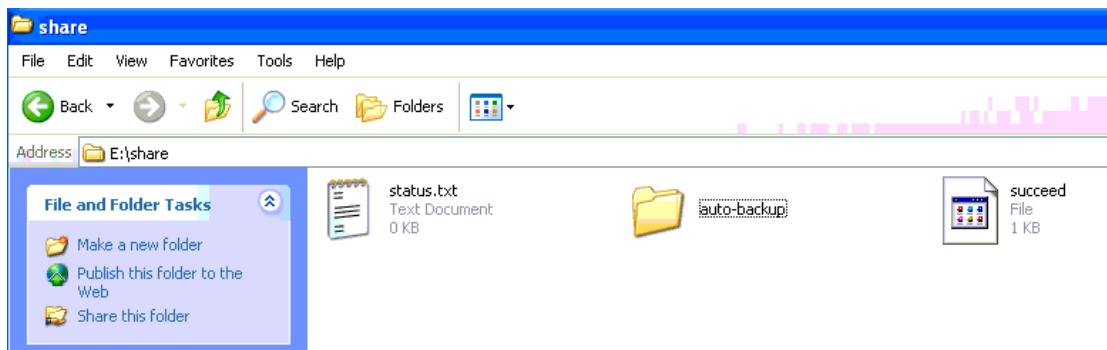


Figure B-3 Autobackup setting succeed

APPENDIX C How to Configure NAT setting

When MyPBX is behind a NAT(firewall),you need to configure NAT setting on MyPBX if you want to use a remote extension.

Please follow section **1** or **2** below depending on your network configuration.

1. If MyPBX is connected to a local network, you must set up port forwarding on your router. Specifically, you must map port 5060 (default SIP port) and port 10001-10200(default RTP port range) as UDP ports.

Next, go to the MyPBX web interface and configurerethe SIP settings according to Figure C-1:

External IP Address: your router's public IP address

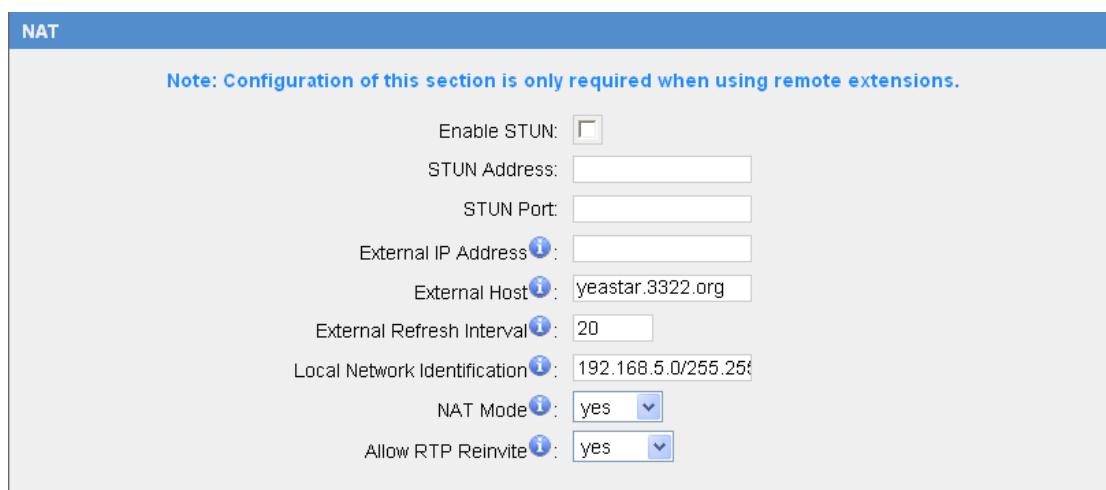
External Host: your router's domain

ExternalRefresh Interval: 20 seconds

Local Network Identification:192.168.5.0/255.255.255.0 (change this according to your network configuration)

NAT mode: Yes

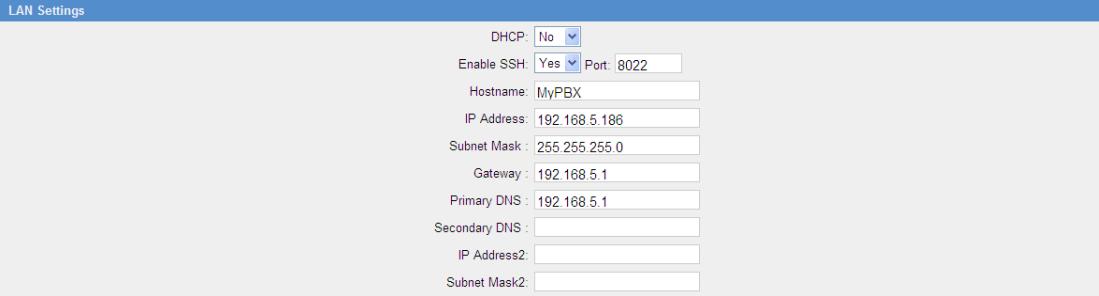
Allow RTP Reinvite: No



NAT	
Note: Configuration of this section is only required when using remote extensions.	
Enable STUN:	<input type="checkbox"/>
STUN Address:	<input type="text"/>
STUN Port:	<input type="text"/>
External IP Address <small>i</small> :	<input type="text"/>
External Host <small>i</small> :	yeastar.3322.org
External Refresh Interval <small>i</small> :	20
Local Network Identification <small>i</small> :	192.168.5.0/255.255.255.0
NAT Mode <small>i</small> :	yes
Allow RTP Reinvite <small>i</small> :	yes

Figure C-1

Assuming that your router's host address is yeastar.3322.org, your local network is from 192.168.5.1-192.168.5.254, and the subnet Mask is 255.255.255.0, the MyPBX network settings should configured like Figure C-2

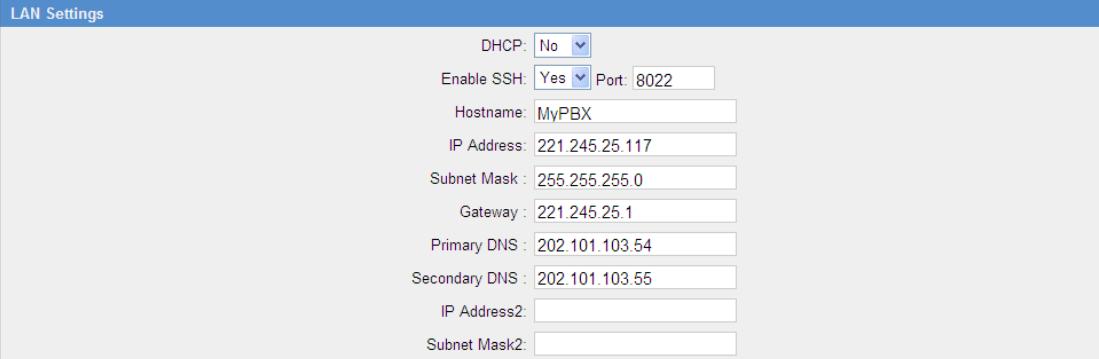


LAN Settings

DHCP:	No
Enable SSH:	Yes Port: 8022
Hostname:	MyPBX
IP Address:	192.168.5.186
Subnet Mask:	255.255.255.0
Gateway:	192.168.5.1
Primary DNS:	192.168.5.1
Secondary DNS:	
IP Address2:	
Subnet Mask2:	

Figure C-2 MyPBX Network setting

2. If MyPBX has a public IP, (i.e. is connected directly to your internet service provider), the network settings should be configured according to Figure C-3:

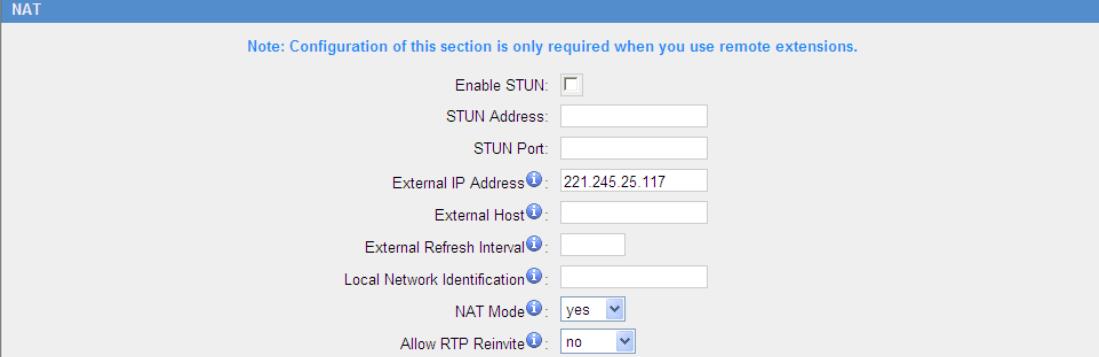


LAN Settings

DHCP:	No
Enable SSH:	Yes Port: 8022
Hostname:	MyPBX
IP Address:	221.245.25.117
Subnet Mask:	255.255.255.0
Gateway:	221.245.25.1
Primary DNS:	202.101.103.54
Secondary DNS:	202.101.103.55
IP Address2:	
Subnet Mask2:	

Figure C-3

Next, you should configure the NAT settings according to Figure C-4



NAT

Note: Configuration of this section is only required when you use remote extensions.

Enable STUN:	<input type="checkbox"/>
STUN Address:	
STUN Port:	
External IP Address <small>(?)</small> :	221.245.25.117
External Host <small>(?)</small> :	
External Refresh Interval <small>(?)</small> :	
Local Network Identification <small>(?)</small> :	
NAT Mode <small>(?)</small> :	yes
Allow RTP Reinvite <small>(?)</small> :	no

Figure C-4

External IP Address: The public IP address of MyPBX

External Host: Leave this blank if no domain has been configured

External Refresh Interval: Leave this blank

Local Network Identification: Leave this blank

NAT mode: Yes

Allow RTP Reinvite: No

APPENDIX D How to Use Auto Provision

Step1. Disable DHCP Server on your local network.
E.g. Disable DHCP Server on Linksys Router.

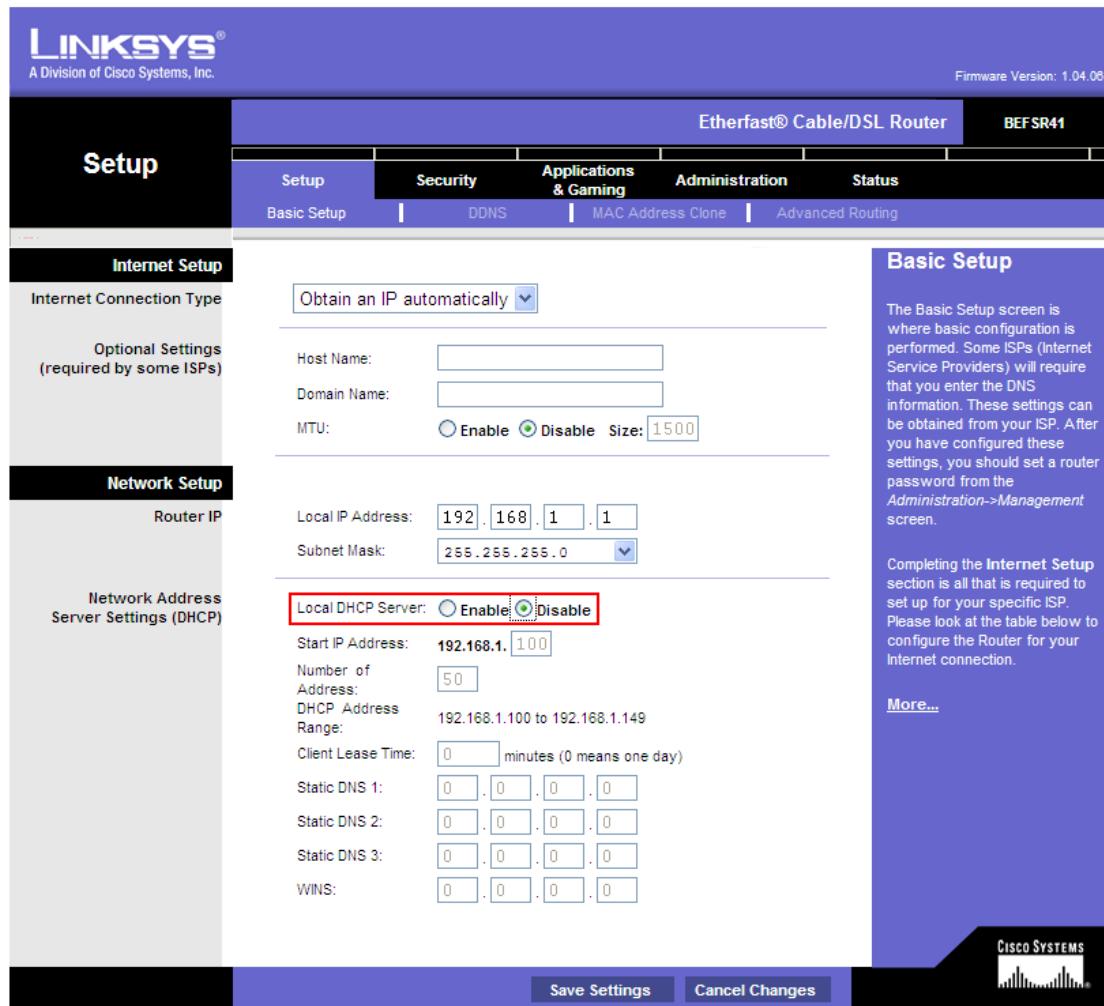


Figure D-1

Step2. Enable DHCP Server on MyPBX.
Login MyPBX web interface, System Settings → DHCP Server → Enable DHCP Server.

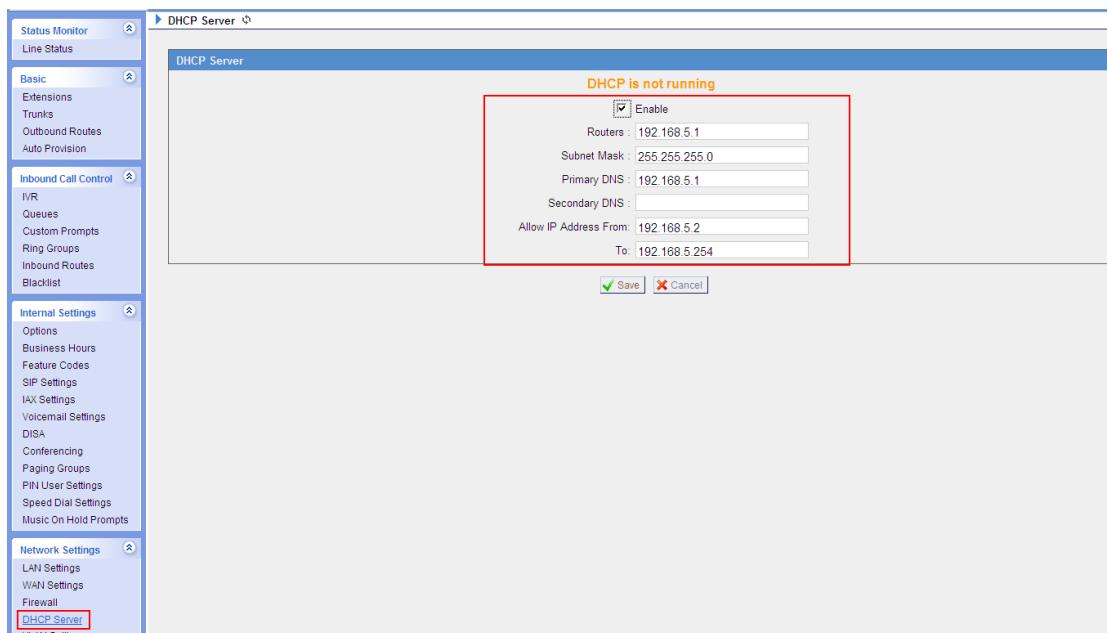


Figure D-2

Step3. Configure phones on MyPBX auto-provision page.

1. Login MyPBX web interface, Basic → Auto Provision → Create New Phone.

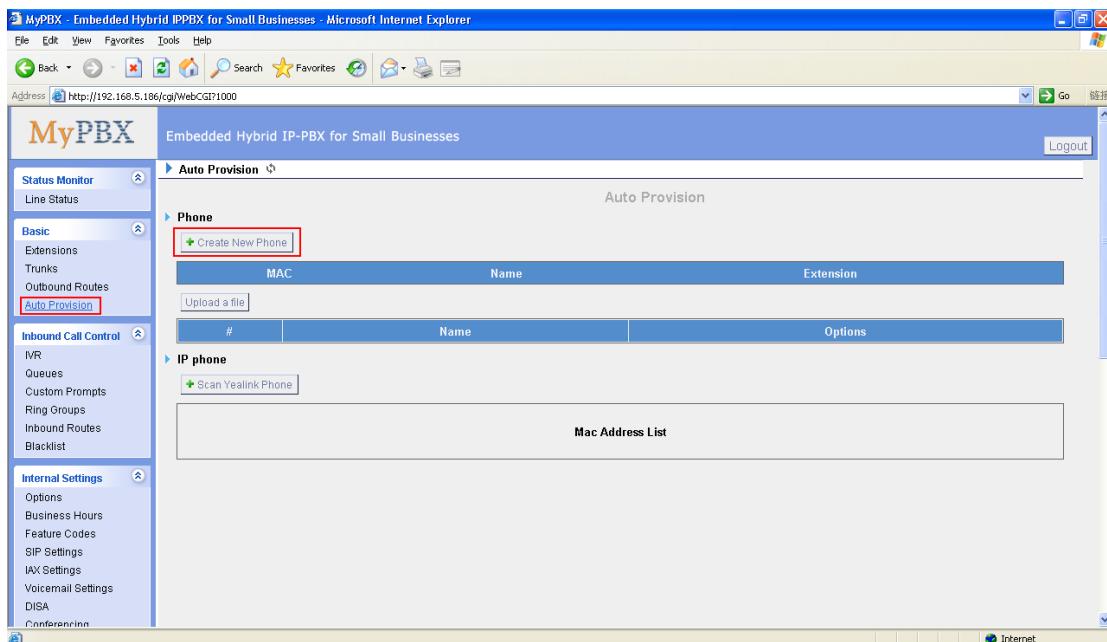


Figure D-3

2. Fill in the phone detail message on the pop-up windows.

Input IP Phone's MAC address, configure Name, Callwaiting, Line, Extension, Label, Line active for the phone.

Create New Phone

General			
MAC:	00156511189E	Name:	Rela
Call Waiting :	Enabled		
Key As Send :	#		
Line			
<input checked="" type="checkbox"/> Line1	Extension: 500	Label: 500	Line Active: <input checked="" type="checkbox"/>
<input type="checkbox"/> Line2	Extension:	Label:	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line3	Extension:	Label:	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line4	Extension:	Label:	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line5	Extension:	Label:	Line Active: <input type="checkbox"/>
<input type="checkbox"/> Line6	Extension:	Label:	Line Active: <input type="checkbox"/>
<input checked="" type="button"/> Save <input type="button"/> Cancel			

Figure D-4

Step4. Turn on the power and connect the network cable to IP Phone.

Remark: The factory default setting of DHCP for IP Phone is enable, so you can skip this step to step 5.

If the DHCP is disable, please follow below step to enable it. (e.g.: Yealink's IP Phone).

1. Login IP phone's web page.
2. Enable DHCP.

Yealink

Status	Account	Network	Phone	Contacts	Upgrade	Security
Internet Port (WAN) PC Port Advanced						
<input checked="" type="radio"/> DHCP <input type="radio"/> Static IP Address					NOTE DHCP The device will acquire its IP address from the DHCP server automatically. Static IP Address Set the IP address, Subnet Mask, Default Router IP address, Primary DNS, Secondary DNS fields manually. PPPoE This setting provide by DSL.	
IP Address Subnet Mask Default Gateway Primary DNS Secondary DNS						
<input checked="" type="radio"/> PPPoE User Password					<input type="button"/> Confirm <input type="button"/> Cancel	

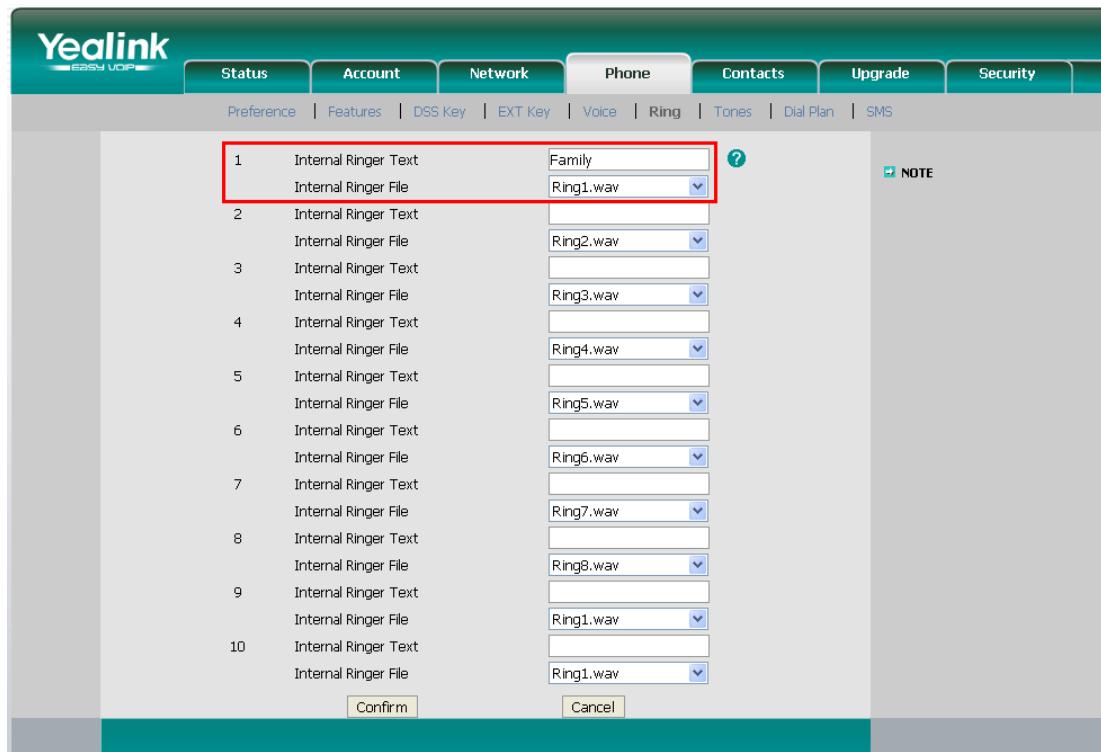
Figure D-5

Step5. Finish.

APPENDIX E HowDo I Configure Distinctive Ring Tones

Step1: On your IP phone, navigate to the Phone settings web configuration page and find the Distinctive Ring Tone section.

For each custom ring tone, enter the Internal Ringer Text (can be digits or text) to trigger the ring tone. For example, you may enter "Family".
e.g.: Yealink's IP phone.



FigureE-1

Step2.Configure the 'Distinctive Ringtone' on MyPBX.

MyPBX web interface, Inbound Routes →Edit Inbound Route, fill in the Internal Ringer Text on 'Distinctive Ringtone'.

Edit Inbound Route: allin

General

Route Name <small>i</small>	: allin
DID Number <small>i</small>	:
Extension <small>i</small>	:
Caller ID Number <small>i</small>	:
Distinctive Ringtone <small>i</small>	: Family

Member Trunks i

Available Trunks	Selected
<input type="button" value=">>"/>	pstn1(FXO)
<input type="button" value="→"/>	pstn2(FXO)
<input type="button" value="←"/>	pstn6(FXO)
<input type="button" value="<<"/>	GSM11(GSM)
	BriTrunk7(BRI)
	BriTrunk8(BRI)
	8032(SIP)
	voipprovider(SIP)

During Office Hours

Destination:	<input type="radio"/> End Call
	<input type="radio"/> Extension
	<input type="radio"/> Voicemail
	<input checked="" type="radio"/> IVR
	<input type="radio"/> RingGroup
	<input type="radio"/> Conference Room
	Extension -- 500
	Voicemail -- 500
	IVR -- welcome
	RingGroup -- ringgroup_defi
	Conference Room -- 640

FigureE-2

Step3. Finish.

APPENDIX F How to Use DID

Direct inward dialing (DID), also called direct dial-in (DDI) in Europe and Oceania, is a feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk.

MyPBX support DID, you can configure DID in inbound route. Related settings: **DID Number, Extension, Destination.**

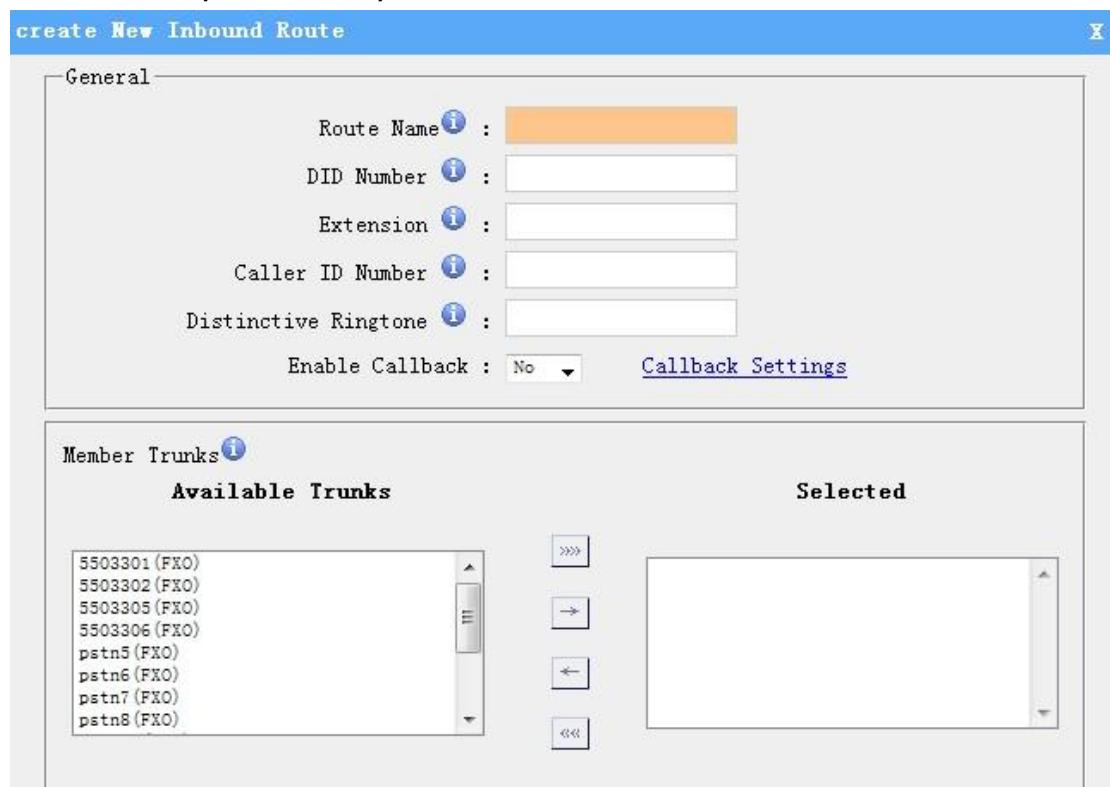


Figure F-1

DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only service provider, E1 trunks, BRI trunks or SIP trunks need to be configured with this setting.

You can also use pattern matching to match a range of numbers. The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9

[12345-9]: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

·Extension

Define the extension for DID number, this field only valid when use E1 trunk for this inbound router. You can only input number and '-' in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

·Destination

If you don't set the extension, you can set the destination of the call here.

Example 1:

Step1: You set the DID number (5503XXX in this example).

Step2: You choose the destination(Ring Group in this example).

The configuration of this example means when the incoming call with DID number 5503XXX (7 digits number start with 5503) will go to the destination Ring Group.

If you choose the destination, please leave the Extension form blank.

create New Inbound Route

General

Route Name  :	<input type="text" value="BRI1"/>
DID Number  :	<input type="text" value="5503XXX"/>
Extension  :	<input type="text"/>
Caller ID Number  :	<input type="text"/>
Distinctive Ringtone  :	<input type="text"/>
Enable Callback :	No <input type="button" value="Callback Settings"/>

Member Trunks 

Available Trunks	Selected
<input type="checkbox" value="5503301 (FXO)"/> 5503301 (FXO) <input type="checkbox" value="5503302 (FXO)"/> 5503302 (FXO) <input type="checkbox" value="5503305 (FXO)"/> 5503305 (FXO) <input type="checkbox" value="5503306 (FXO)"/> 5503306 (FXO) <input type="checkbox" value="pstn5 (FXO)"/> pstn5 (FXO) <input type="checkbox" value="pstn6 (FXO)"/> pstn6 (FXO) <input type="checkbox" value="pstn7 (FXO)"/> pstn7 (FXO) <input type="checkbox" value="pstn8 (FXO)"/> pstn8 (FXO)	<input type="button" value=">>>"/> <input type="button" value=">"/> <input type="button" value="<"/> <input type="button" value="<<"/> <input type="button" value="<<<"/>

During Office Hours

<input type="radio"/> End Call	
<input type="radio"/> Extension	<input type="button" value="Extension -- 500"/>
<input type="radio"/> Voicemail	<input type="button" value="Voicemail -- 500"/>
<input type="radio"/> IVR	<input type="button" value="IVR -- welcome"/>
<input checked="" type="radio"/> RingGroup	<input type="button" value="RingGroup -- NationalSe"/>
<input type="radio"/> Conference Room	<input type="button" value="Conference Room -- 640"/>
<input type="radio"/> DISA	<input type="button" value="DISA -- test"/>
<input type="radio"/> Queues	<input type="button" value="Queues -- test"/>
<input type="radio"/> Faxes 	<input type="button" value="Faxes -- 500"/>
<input type="radio"/> Outbound Routes 	<input type="button" value="Route Name -- toOCS"/>

Figure F-2

Example 2:

Step1: You set the DID number (6001-6099 in this example).

Step2: You set the Extension (6001-6099 in this example).

The configuration of this example means when the incoming call with DID number 6001 to 6099 will go to the destination 6001 to 6099(number 6001 to extension 6001, number 6002 to extension 6002).

The destination you set below will be disabled if you set the Extension.

create New Inbound Route

General

Route Name <small>i</small>	: BRI1
DID Number <small>i</small>	: 6001-6099
Extension <small>i</small>	: 6001-6099
Caller ID Number <small>i</small>	:
Distinctive Ringtone <small>i</small>	:
Enable Callback	: No <input type="button" value="Callback Settings"/>

Member Trunks i

Available Trunks	Selected
5503301 (FXO)	
5503302 (FXO)	
5503305 (FXO)	
5503306 (FXO)	
pstn5 (FXO)	
pstn6 (FXO)	
pstn7 (FXO)	
pstn8 (FXO)	

Buttons: >>> > < <<<

Figure F-3

APPENDIX G MyPBX Security configuration Guide

VoIP attacks, although it is not an everyday occurrence, it does exist. While using VoIP, system security is undoubtedly one of the issues we care about most. But with the appropriate configuration, and some basic safety habits, we can improve the security of the telephone system. Moreover, the powerful built-in firewall function in MyPBX is adequate to enable the system to run safely and stably.

This guide will introduce the highest defense level in MyPBX, and we strongly recommend that you configure firewall and other security options according to this guide, to prevent the attack fraud and the system failure or calls loss.

Note: In this guide, the configuration options marked with '*' only exist in 12.17.XX.XX and above versions, namely, 3.2 guest calls option, 3.3 remote registered option, and 5 alarm settings.

1. Security Configuration for Web GUI

1.1 Change the default access port for HTTP on Options page

PBX→Basic Settings→General Preferences→HTTP Bind Port

General Preferences

General Preferences

Ring Timeout	30	s
MAX Call Duration	6000	s
Maximum Concurrent Calls	0	
Music On Hold:	calmriver	
Tone Region	United States/North America	
HTTP Bind Port	80	
FXO Mode	FCC	
Virtual Ring Back Tone	No	
Distinctive Caller ID	No	
Follow Me Prompt	Yes	
Music on hold for Follow Me	Default	
Invalid Phone Number Prompt		
Busy Line Prompt		
Dial Failure Prompt		

Figure G-1

1.2 Change the default password for the web GUI

System→System Preferences→Change Password

Change Password

User:	admin
Enter Old Password:	
Enter New Password:	
Retype New Password:	

FigureG-2

2. Disable SSH on LAN Settings Page

2.1 Disable SSH

Select LAN Settings→Enable SSH. If external debugging isn't required, please select "No".

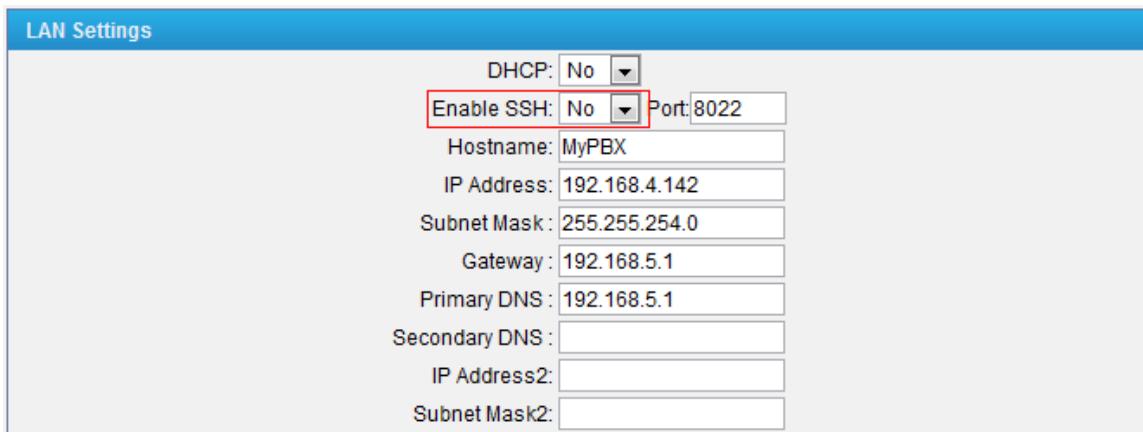


Figure G-3

2.2 Change the default password for SSH

We can use the Linux command **passwd** to change root password of MyPBX.

1. Login via putty.exe

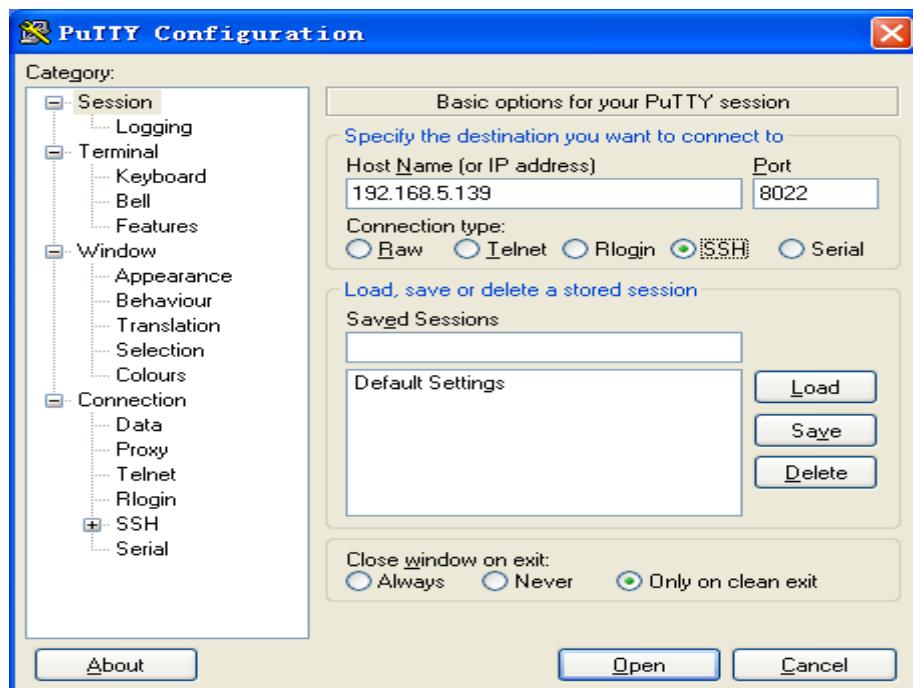


Figure G-4

2. The default username is **root** and the default password is **ys123456**.

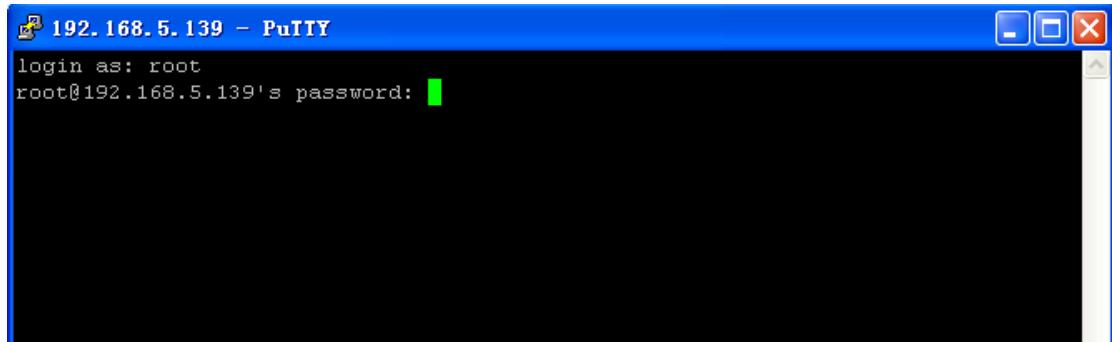


Figure G-5

3. Step 2 use command **passwd** to change the root's password

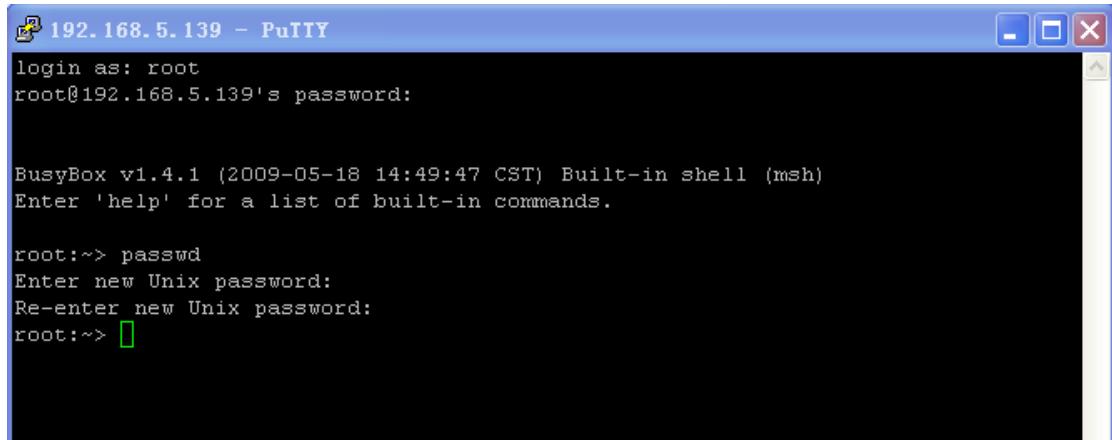


Figure G-6

3. Security Configuration for Extensions

3.1 Change the default SIP Port

PBX→Basic settings→SIP Settings→General→UDP Port

SIP Settings

General NAT Codecs QOS Advanced Settings

<input checked="" type="checkbox" value="Enable"/>	UDP Port	5060
<input checked="" type="checkbox" value="Enable"/>	TCP Port	5060
<input checked="" type="checkbox" value="Enable"/>	TLS Port	5061
	RTP Port Start	10000
	RTP Port End	12000
	DTMF Mode	rfc2833
	Max Registration/Subscription Time	3600
	Min Registration/Subscription Time	60
	Default Incoming/Outgoing Registration Time	120
	Register Attempts	8
	Register Timeout	20
	Calling Channel Codec Priority	Yes
	Video Support	Yes
	Max Bit Rate	384 kb/s
	DNS SRV Look Up	No
	User Agent	

Figure G-7

3.2*Disable guest calls

PBX→Basic settings→SIP Settings→Advanced Settings→Allow Guest

SIP Settings

General NAT Codecs QOS Advanced Settings

From Field:	From
To Field:	INVITE
180 Ringing:	<input type="checkbox"/>
Remote Party ID	<input type="checkbox"/> send <input type="checkbox"/> trust
Allow Guest	No
Pedantic	No
Session-timers	Accept
Session-expires	1800 s
Session-minse	90 s
Session-refresher	Uas

Figure G-8

3.3* Security Configuration for remote extensions

If remote registration isn't required, please disable it.

PBX→Extensions→FXS/VoIP Extensions→VoIP Extensions→General→VoIP Settings

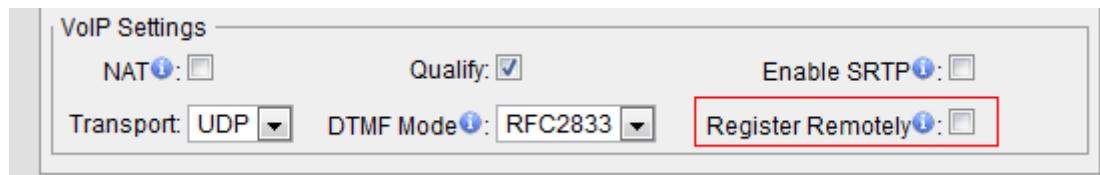


Figure G-9

3.4 Set an enhanced password and enable IP restriction for extensions

1) Set a new extension password at the higher security level, e.g. AjK5Up1G.

PBX→Extensions→FXS/VoIP Extensions→VoIP Extensions→General→Password

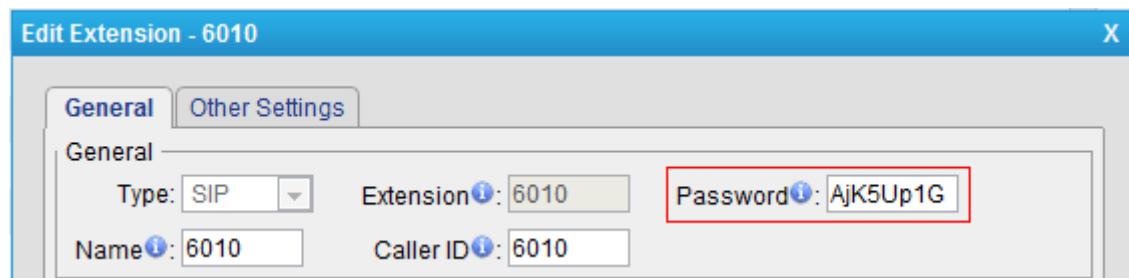


Figure G-10

2) Enable IP restriction and enter the permitted "IP address/Subnet mask", e.g. 192.168.5.136.

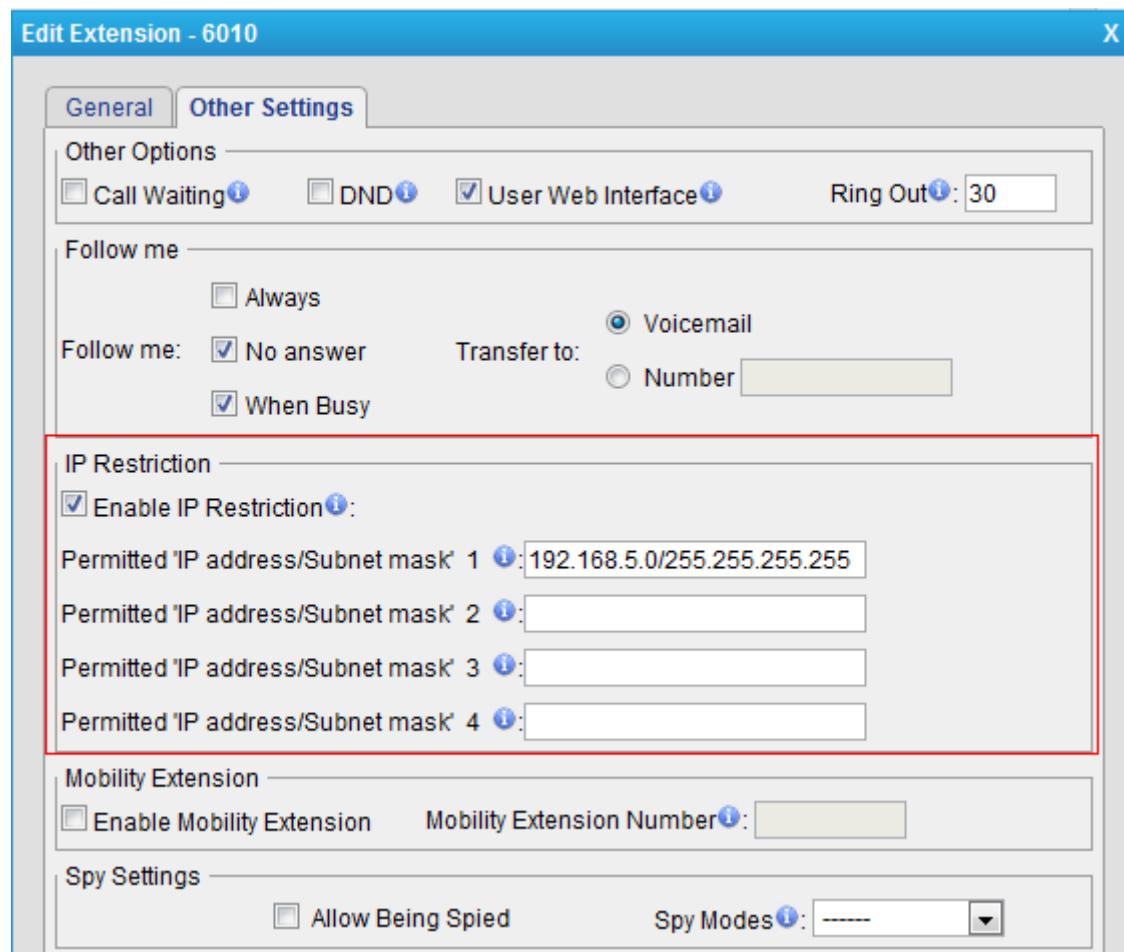


Figure G-11

4. Setup Proper Firewall Rules

Note: Please backup the configurations on backup and restore page before you go ahead. In the case that you lock the device, you can reset to factory default and restore the previous configurations. Below example rules works with MyPBX firmware version 2.15.xx.xx or higher versions.

Step1. Enable firewall on firewall page of MyPBX.
 System→Firewall Settings→Firewall Rules→General Settings

General Preferences

General Settings

Note:

1. You must reboot the system after enabling or disabling firewall.
2. It is strongly recommended to add local network address to a common rule with the 'action' is 'accept', or it may be dragged into the blacklist.

Enable Firewall

Disable Ping

Drop All

Firewall is not enabled

Figure G-12

Step2. Add a common rule to accept local network access.

Create a common rule to allow all the IP addresses of the local phones to access MyPBX server. For example, if the IP addresses of the local network are 192.168.5.1-254, the configurations could be as below:

Name: LocalNetwork

Protocol: BOTH

Port: 1:65535

IP: 192.168.5.0/255.255.255.0

Action: Accept

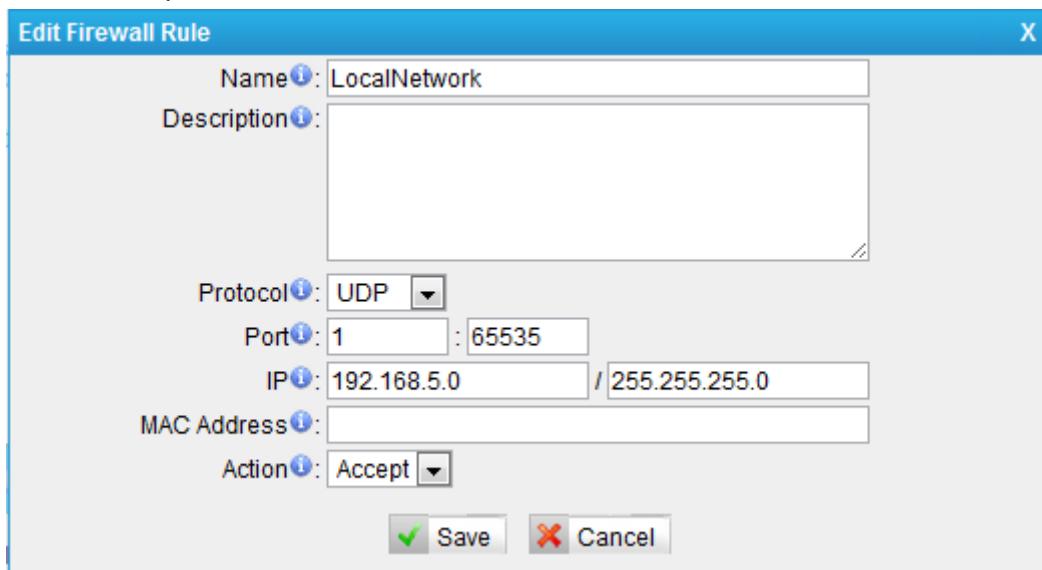


Figure G-13

Step3. Create common rules to accept remote extensions or remote administrators, if you use SIP trunk, please accept the provider's host as well.

Note: If there are no remote extensions, the rule is not required.

- 1) Set up the firewall rule to allow the public IP address of remote extensions to access MyPBX server. e.g.110.111.132.6, the configurations could be as below:

Name: Remote Extension

Protocol: BOTH

Port: 1:65535

IP: 110.111.132.6/255.255.255.255

Action: Accept

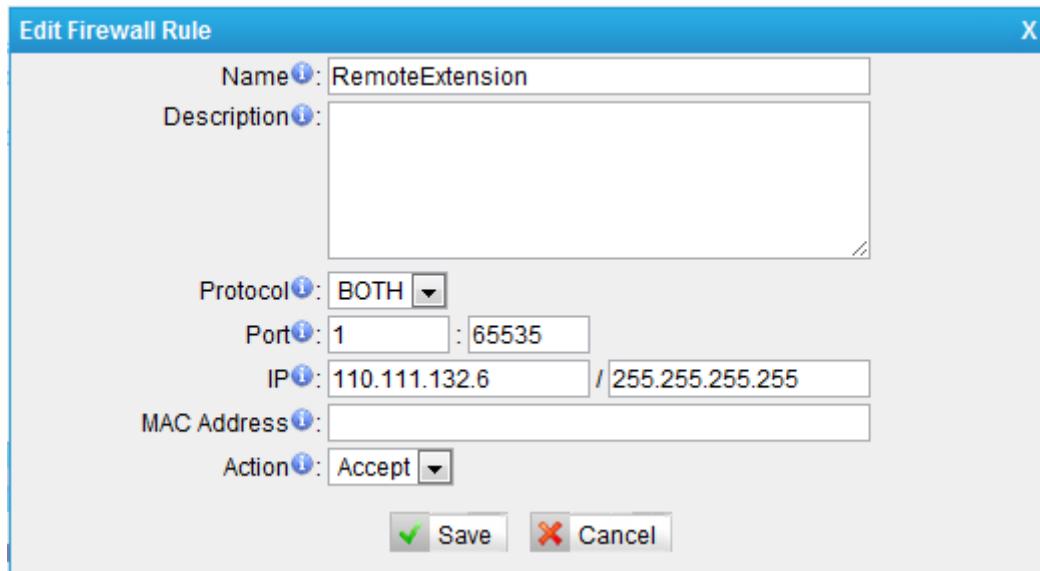


Figure G-14

Step4. Configure auto blacklist rules

Auto blacklist rules: the Server would add the IP address to the blacklist automatically if the number of the packets it sends exceed the rule you configured.

- 1) Add two auto blacklist rules for port: 5060.

Rule No.1:

Port: 5060

Protocol: UDP

IP Packets: 120

Time Interval: 60 seconds

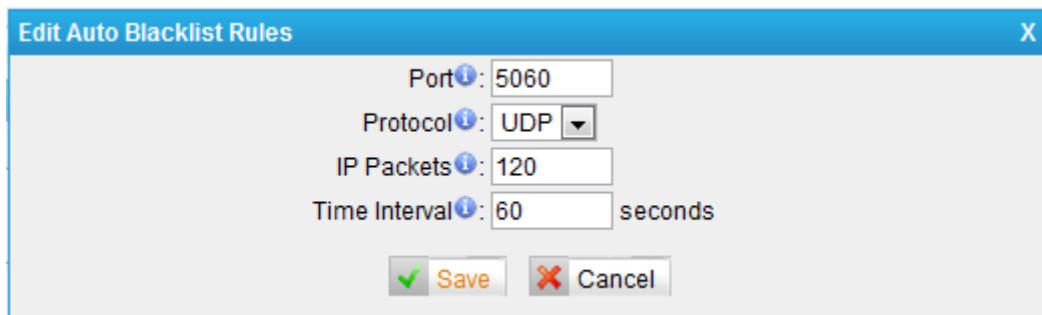


Figure G-15

Rule No.2:

Port: 5060

Protocol: UDP

IP Packets: 40

Time Interval: 2 seconds

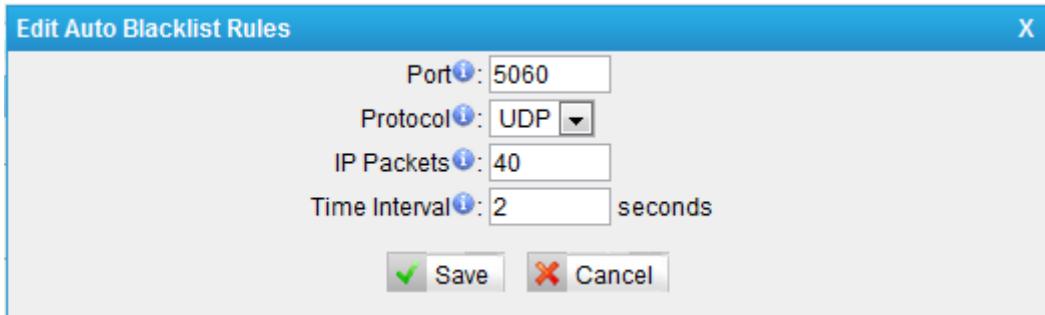


Figure G-16

2) Add an auto blacklist rule for Port:8022

Rule No.3

Port: 8022

Protocol: TCP

IP Packets: 5

Time Interval: 60 seconds

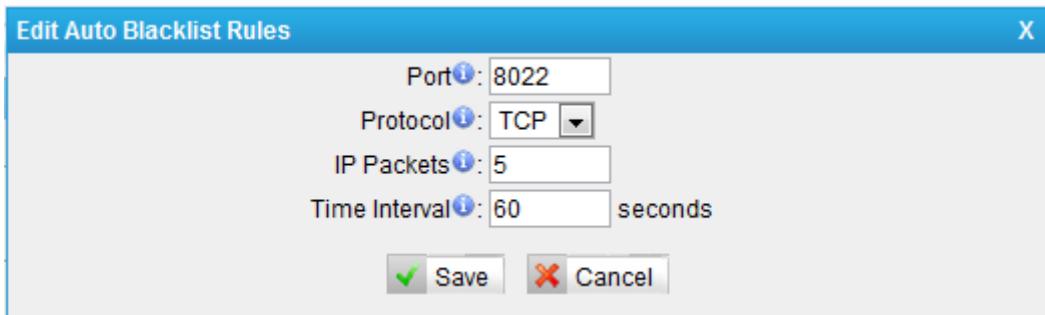


Figure G-17

Step 5. Add a Firewall Rule for VoIP trunk registration

Note: If there is no VoIP trunk, this rule is not required. And if the RTP IP address of VoIP trunk and Registration IP address of the VoIP trunk are different, we need create a rule to accept the RTP IP address too.

Add a rule to accept the IP address of the VoIP trunk to access MyPBX server. For example: If the IP address of the VoIP trunk is 110.5.14.6, Protocol is UDP and Port is 5060, the configuration could be as below:

Name: VoIPTrunk

Protocol: UDP

Port: 5060: 5060

IP: 110.5.14.6/255.255.255.255

Action: Accept

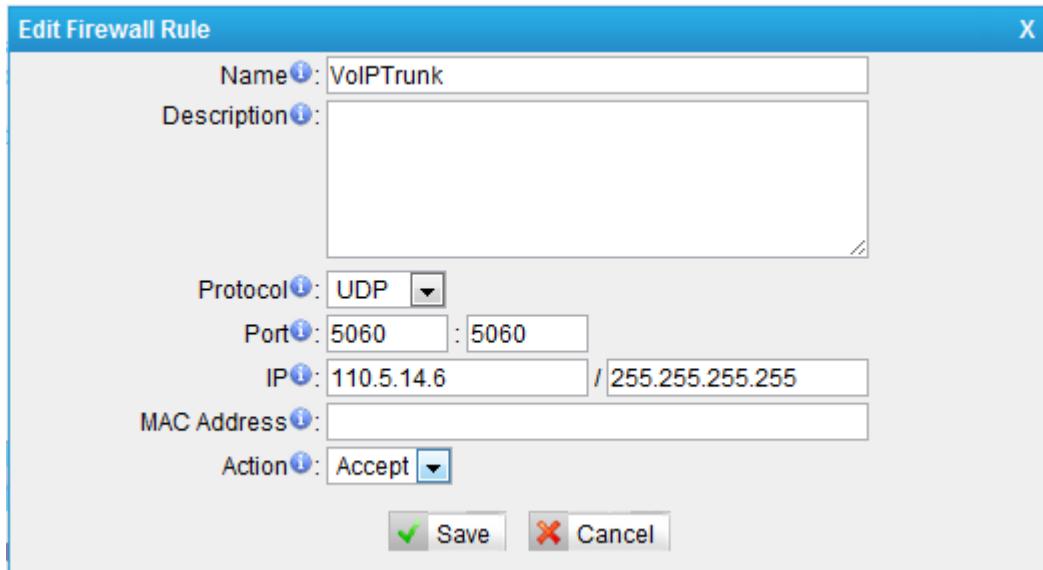


Figure G-18

Step6. Add a firewall rule to accept the remote access of HTTP port. For example, if the remote access IP is 110.5.14.6, and the port is 80, the configuration could be as below.

Name: RemoteHTTP

Protocol: TCP

Port: 80:80

IP: 110.5.14.6/255.255.255.255

Action: Accept

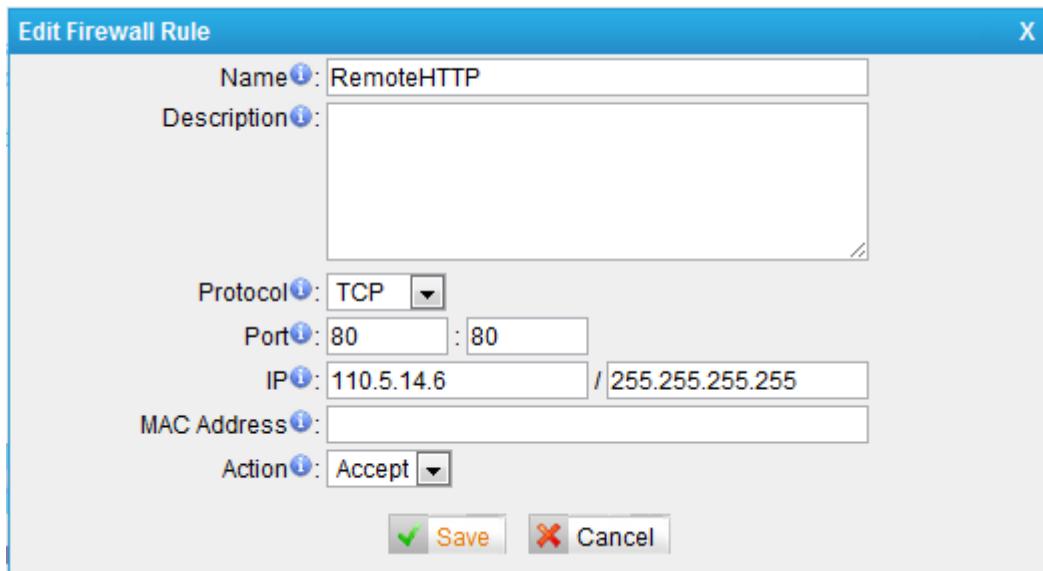


Figure G-19

Step 7. Add a firewall rule to accept remote access of SSH port. For example: if the remote access IP is 110.5.14.6 and the port is 8022.

Note: If the remote access of SSH port is not needed, this rule is not required.

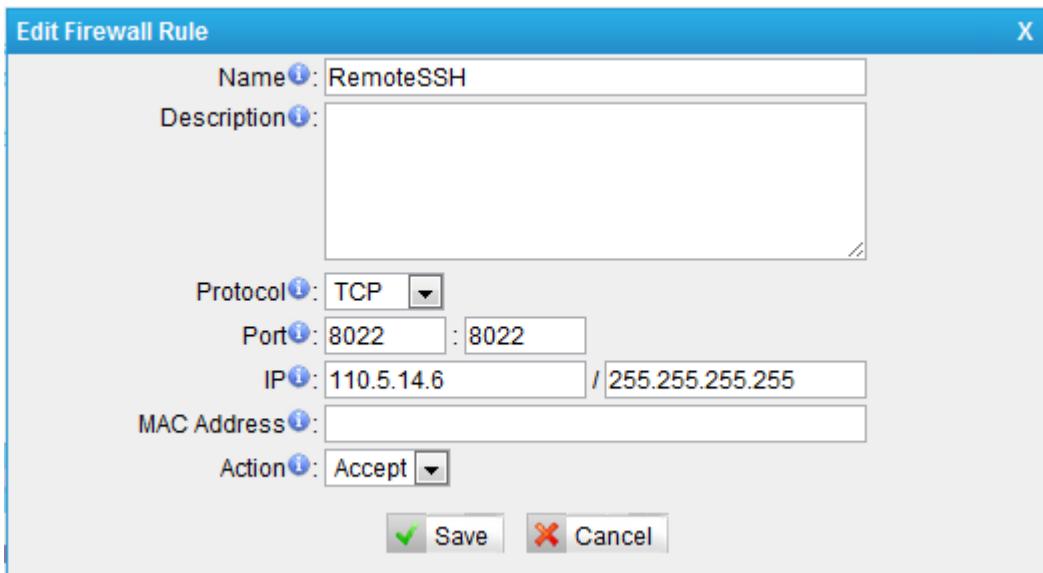
Name: RemoteSSH

Protocol: TCP

Port: 8022:8022

IP: 110.5.14.6/255.255.255.255

Action: Accept



The dialog box is titled "Edit Firewall Rule". It contains the following fields:

- Name:** RemoteSSH
- Description:** (empty)
- Protocol:** TCP
- Port:** 8022 : 8022
- IP:** 110.5.14.6 / 255.255.255.255
- MAC Address:** (empty)
- Action:** Accept

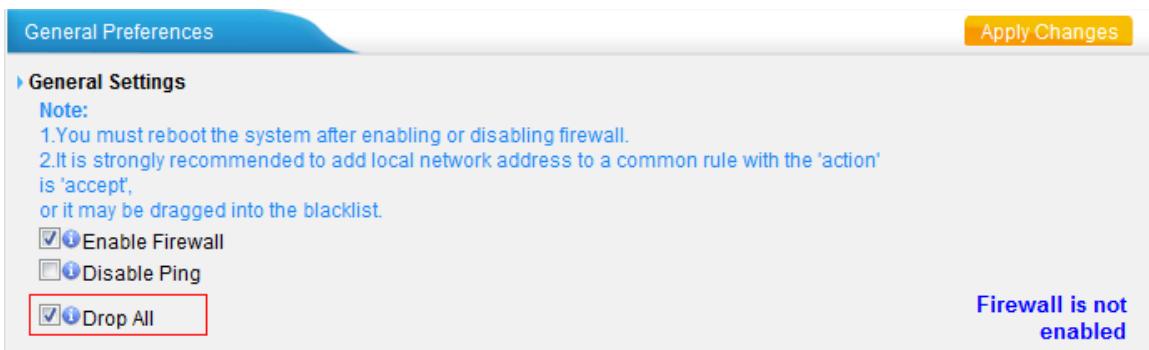
At the bottom are "Save" and "Cancel" buttons.

Figure G-20

Step 8. Add other firewall rules by yourself. For example, if you are using features about email, you should add the firewall rules for the SMTP server and POP3 server.

Step 9. Enable Drop all (If this feature is enabled, all the packets and connection that do not match the rules would be dropped.)

Note: Before enable this feature, please add a rule to accept the local network access, or the server might not be accessed.



The screen shows the "General Preferences" tab selected. Under "General Settings", there is a note and three checkboxes:

- Note:**
 - You must reboot the system after enabling or disabling firewall.
 - It is strongly recommended to add local network address to a common rule with the 'action' is 'accept', or it may be dragged into the blacklist.
- Enable Firewall
- Disable Ping
- Drop All

A message "Firewall is not enabled" is displayed next to the "Drop All" checkbox.

Figure G-21

Step 10. The Configuration of firewall settings is completed. See as below figure.

The screenshot shows the 'General Preferences' section with the 'Firewall' tab selected. Under 'General Settings', there is a note about rebooting after enabling or disabling the firewall. Three checkboxes are present: 'Enable Firewall' (checked), 'Disable Ping' (unchecked), and 'Drop All' (checked). A blue message box on the right states 'Firewall is not enabled'. The 'Common Rules' section contains a table with the following data:

Action	Name	Protocol	IP	MAC Address	Port
ACCEPT	LocalNetwork	UDP	192.168.5.0/255.255.255.0	--	1:65535
ACCEPT	RemoteExtension	BOTH	110.111.132.6/255.255.255.255	--	1:65535
ACCEPT	VoIPTrunk	UDP	110.5.14.6/255.255.255.255	--	5060:5060
ACCEPT	RemoteHTTP	TCP	110.5.14.6/255.255.255.255	--	80:80
ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	--	8022:8022

The 'Auto Defense' section shows a table with the message 'No Auto Defense Rules Defined'.

Figure G-21

The screenshot shows the 'IP Blacklist' section. Under 'Blacklist Rules', there is a table with the following data:

Port	Protocol	Rate
5060	UDP	120/60s
5060	UDP	40/2s
8022	TCP	5/60s

The 'IP Blacklist' section shows a table with the message 'No Auto Black IP Address'.

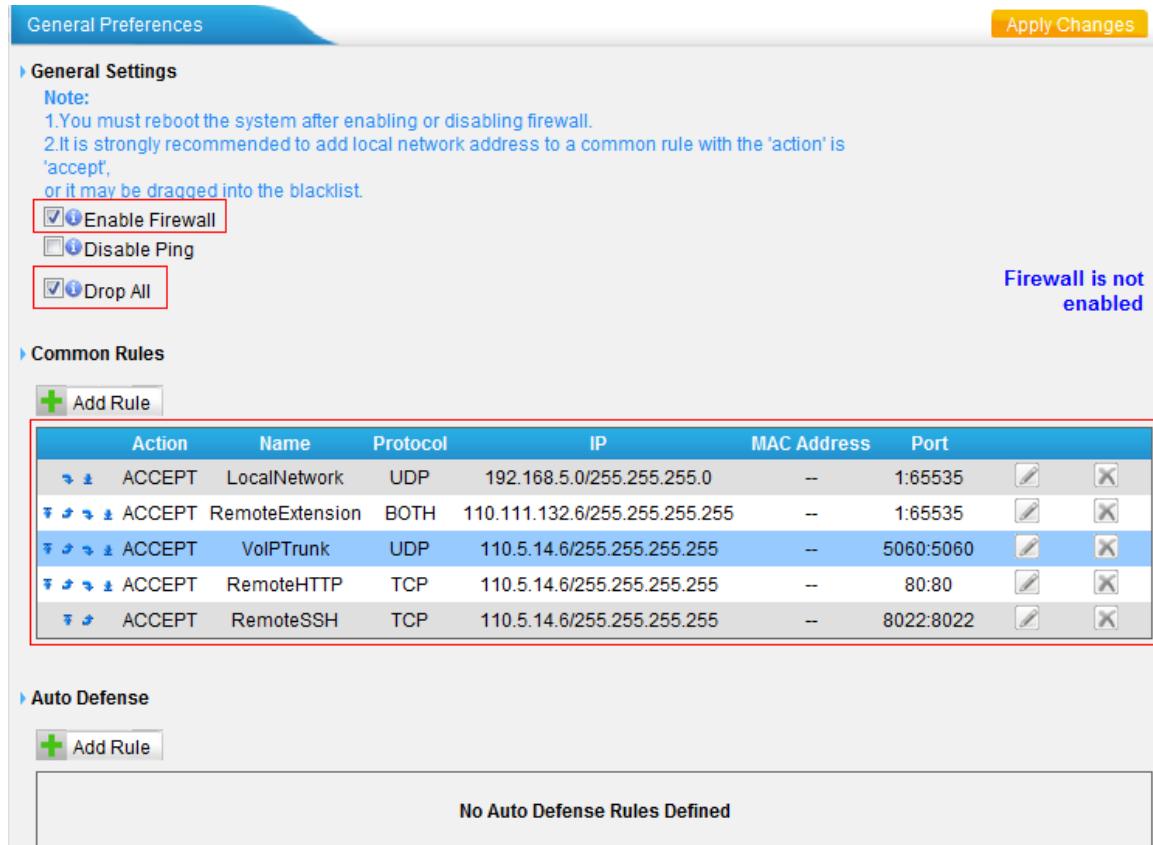
Figure G-22

Step 11. Define the authorized data transmitting rate and drop all other unauthorized hosts.

Note: If we setup IP blacklist rules here, those unauthorized hosts will still be able to send SIP packets into MyPBX if the packets transmitting rate is under the rate defined. But as soon as the transmitting beyond the define rate, the IP will be added into blacklist automatically. This is useful for remote extension who is using dynamic IP address.

Remove all the IP blacklist rules if you don't allow any other unauthorized hosts access MyPBX. See as below figure.

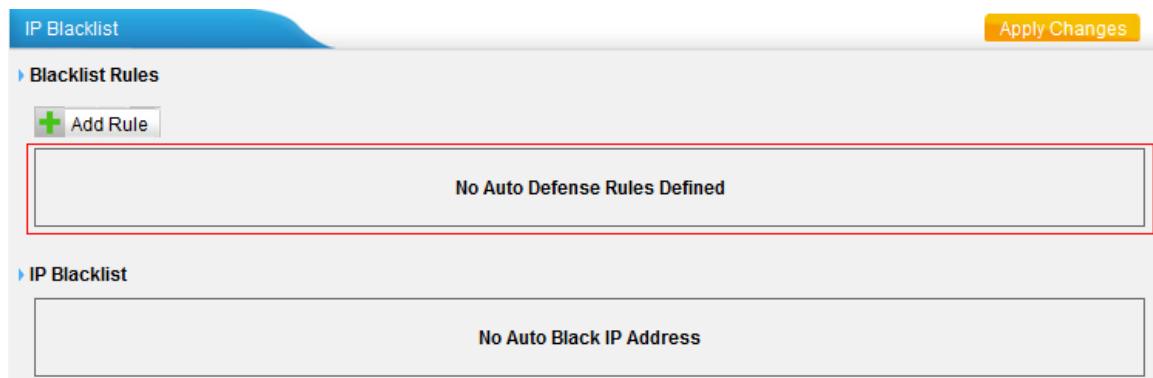
Note: If 'drop all' is not enabled, please don't remove the IP blacklist rules in case the system security hole.



The screenshot shows the 'General Preferences' section of the MyPBX U300 administrator interface. In the 'General Settings' tab, under 'Firewall' settings, the 'Drop All' checkbox is checked and highlighted with a red border. A note on the right states 'Firewall is not enabled'. In the 'Common Rules' tab, a table lists several ACCEPT rules for LocalNetwork, RemoteExtension, VoIPTrunk, RemoteHTTP, and RemoteSSH. The 'Drop All' rule is also listed here. Below the table, the 'Auto Defense' section shows a table with no defined rules.

Action	Name	Protocol	IP	MAC Address	Port
ACCEPT	LocalNetwork	UDP	192.168.5.0/255.255.255.0	--	1:65535
ACCEPT	RemoteExtension	BOTH	110.111.132.6/255.255.255.255	--	1:65535
ACCEPT	VoIPTrunk	UDP	110.5.14.6/255.255.255.255	--	5060:5060
ACCEPT	RemoteHTTP	TCP	110.5.14.6/255.255.255.255	--	80:80
ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	--	8022:8022

Figure G-23



The screenshot shows the 'IP Blacklist' section of the MyPBX U300 administrator interface. Under 'Blacklist Rules', there is a table with no defined rules. Under 'IP Blacklist', there is a table with no defined rules.

No Auto Defense Rules Defined		

No Auto Black IP Address		

Figure G-24

5*. Alert Settings

After enabling alert settings', if the device is attacked, the system will notify users the alert via call or e-mail. The attack modes include IP attack and Web Login.

5.1 IPATTACK

When the system is attacked by some IP addresses, the firewall will add the IP to auto IP Blacklist and notify the user if it match the protection rule.

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

Phone Notification Settings:

Phone Notification: Yes

Number: 500;5503301

Attempts: 1

Interval: 60s

Prompt: default

Note: If there's outbound number to notify, the number should be fit with the dial pattern of outbound route.

E-mail Notification Settings:

E-mail Notification: Yes

To: alert@yeastar.com

Subject: IPAttack

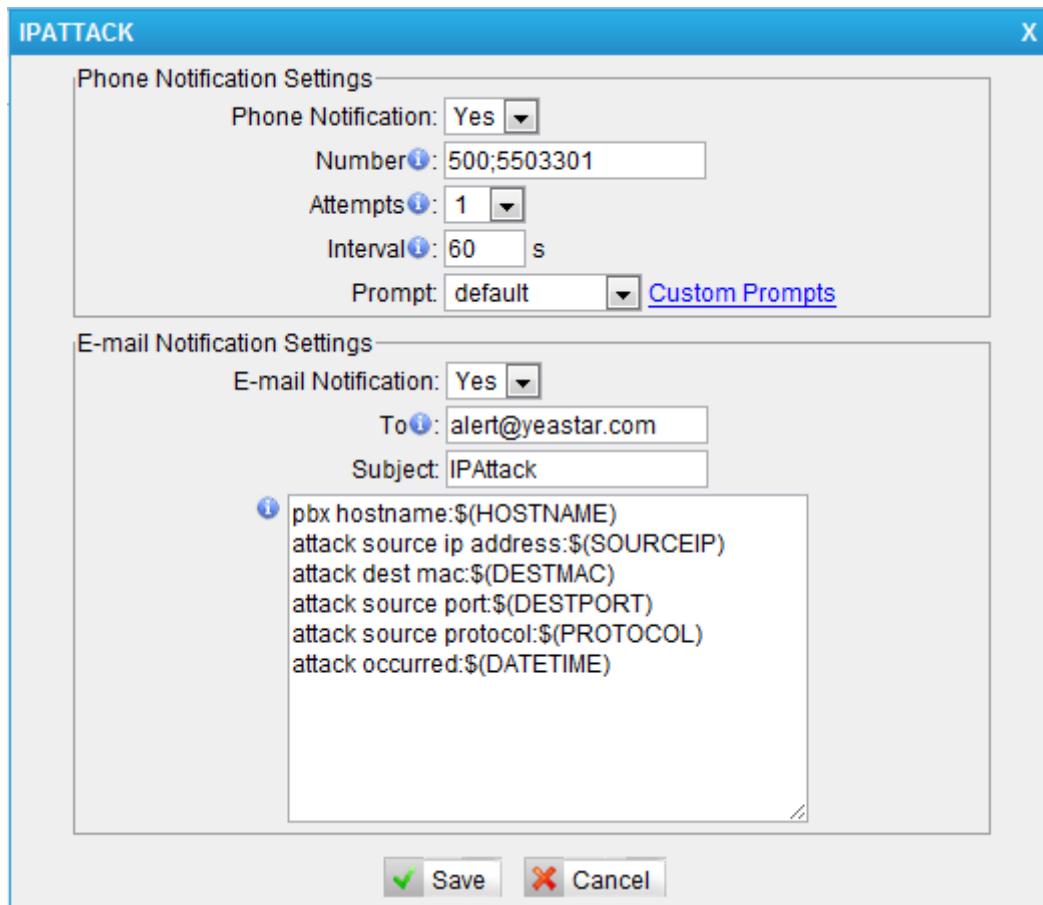


Figure G-25

5.2 WEBLOGIN

Enter the password incorrectly five times when logging in MyPBX Web interface will be deemed as attack, the system will limit the IP login within 10 minutes and notify the user.

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

Phone Notification Settings:

Phone Notification: Yes

Number: 500;5503301

Attempts: 1

Interval: 60s

Prompt: default

Note: If there's outbound number to notify, the number should be fit with the dial pattern of outbound route.

E-mail Notification Settings:

E-mail Notification: Yes

To: alert@yeastar.com

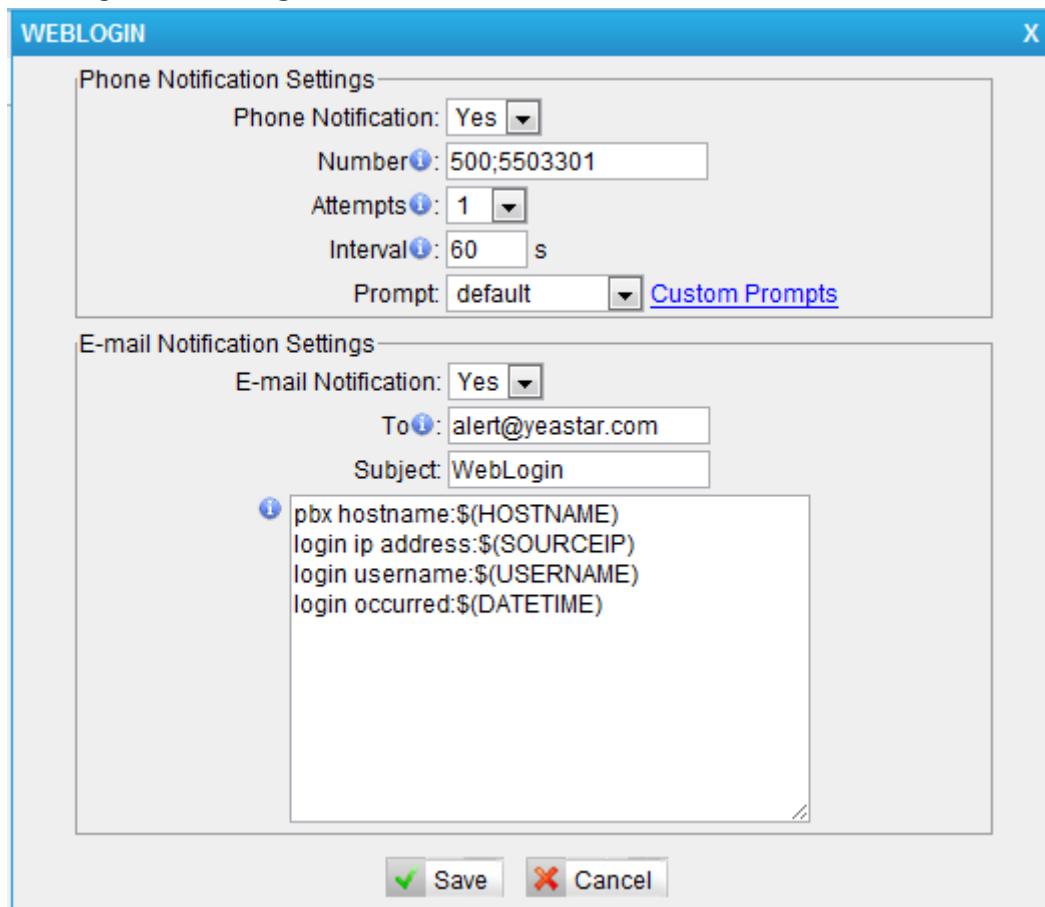
Subject: WebLogin

Figure G-26

6. Note

If the phenomena of toll fraud have been happened in your MyPBX system. We are really sorry about that, then please enhance the protection level of your firewall refer to the above steps.

In addition, please change the all password: Web GUI password, SSH password, and all extensions password.

<Finish>